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A STUDY AND EXPERIMENT PLAN FOR DIGITAL MOBILE COMMUNICATION VIA SATELLITE

FINAL REPORT

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Prepared for:

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AMES RESEARCH CENTER
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A STUDY AND EXPERIMENT PLAN FOR DIGITAL MOBILE COMMUNICATION VIA SATELLITE

- FINAL REPORT -

by

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1.0 INTRODUCTION

The principal objective of this study is to investigate the application of digital communications technology to provide mobile voice communications via a satellite repeater for public service uses. These services include fire control, medical emergencies, civil disturbances, disaster control, and general public agency remote, rural personal speech communication. A large user population is envisioned consisting mainly of small mobile ground terminals, principally land vehicles and portable manpacks, with a few base stations. The study focuses on the needs of the individual user of a small mobile land terminal primarily requiring a single voice channel and secondarily a low-rate data capability, possibly facsimile.

Table 1.0-1 presents the basic assumptions underlying the study and experiment plan for digital mobile communications via satellite (DMCVS) as obtained from the project statement of work. As indicated, the study examines the viability of mobile communications within the context of a frequency division multiple access (FDMA), single channel per carrier (SCPC) satellite system emphasizing digital techniques to serve a large population of users. The intent is to provide these mobile users with a grade of service consistant with the requirements for remote, rural (perhaps emergency) voice communications, but which approaches toll quality speech.

In order to serve the large number of users with a limited satellite channel capacity, use of a demand assignment function is necessary. This requirement is examined within the framework of a centralized scheme using a random access mode for service requests and a common broadcast mode for channel assignments and network control.

The central issues and constraints driving the DMCVS investigation are in brief:

- Applicability of digital voice modulation
- Low-cost land mobile terminals
- Simple terminal operation with fixed antenna
- Public agency and emergency services uses
- System flexibility and growth

Table 1.0-1. Study Assumptions

ITEM	SPECIFICATION
• SATELLITE ANTENNA COVERAGE	CONUS USING ONE OR SEVERAL BEAMS
• MOBILE ANTENNA	FIXED, HEMISPHERICAL COVERAGE
• FREQUENCY ALLOCATIONS:	
MOBILE - SATELLITE - MOBILE	UHF (ABOUT 850 MHz), LAND MOBILE OR L-BAND AERONAUTICAL
MOBILE - CENTRAL STATIONS	UNDEFINED (POSSIBLE Ku-BAND)
• CHANNEL DEFINITION	FREQUENCY DIVISION MULTIPLE ACCESS, SINGLE CHANNEL PER CARRIER
• USER POPULATION	HUNDREDS OR THOUSANDS
• EARTH TERMINAL	SMALL, LOW-COST, LAND MOBILE
• SERVICE TYPES	SINGLE CHANNEL VOICE OR DATA
• GRADE OF SERVICE	REMOTE, RURAL, EMERGENCY
• OPERATIONAL FEATURES	SIMPLE, ALLOWS UNTRAINED USERS
• DATA CHANNEL	LOW-RATE (300 bps - 2.4 kbps), BER $< 10^{-5}$
• SATELLITE TRANSPONDER	FREQUENCY TRANSLATION REPEATER WITHOUT REGENERATION OR SWITCHING
• DEMAND ASSIGNMENT	CENTRALIZED, RANDOM ACCESS REQUEST, BROADCAST CHANNEL ASSIGNMENT
• NETWORK CONNECTIONS	MOBILE TO MOBILE AND MOBILE TO EXISTING NETWORK VIA ONE OR MORE CENTRAL STATIONS

- Use of SCPC - FDMA configuration
- Provision for DAMA with low probability of blocking
- Graceful degradation of digital techniques
- LSI cost-reduction implications
- Low development risk for mobile terminals
- Compatibility of digital voice, data, and DAMA

Macrosopically, the goals of the DMCVS study are met by a three-pronged approach consisting of:

a. An investigation and tradeoff to select the best combination of voice digitization technique, digital modulation technique, and demand access approach.
b. A functional description and cost estimate of an operational DMCVS system employing the selected techniques.
c. The development and planning of a field demonstration experiment and the associated cost for verifying and evaluating the selected approach for mobile communications using an existing NASA experimental communications satellite.

The results of carrying out the DMCVS study as described here are summarized and highlighted in the executive summary contained in Section 2. The remainder of the report describes in detail each of the major constituent study elements.

In order to determine the sizing of the communication satellite system some measure of the expected traffic utilization is necessary. Section 3 derives a traffic model on which to base the determination of the required maximum number of satellite channels to provide the anticipated level of service as a function of the number of mobile terminals and the blocking probability.

Various voice digitization techniques are described and compared in Section 4 including a discussion of speech performance measures, such as signal-to-noise ratio (SNR) intelligibility scores, and voice quality. A point is

made that voice SNR is not a good descriptor for mobile voice communication systems, characterized by low power levels, and that for this application better performance measures are speech intelligibility and quality.

A number of candidate digital modulation schemes are reviewed in Section 5 and compared on the basis of theoretical bit error rate (BER) performance. Also reviewed is the application and implications of forward error correction coding for a mobile voice communication satellite environment.

Section 6 includes a general link analysis of the UHF digital mobile communication system together with parameter variations and evaluations, providing a broad review of systems and techniques before selecting a few candidates. A number of ancillary topics including intermodulation crossproducts, voice-activated carriers, and mobile terminal antennas are addressed in support of this investigation.

Demand assignment multiple access considerations and analysis tradeoffs are presented in Section 7 to provide a basis for justifying the preselection of a centralized DAMA approach using random access for service requests and common mode broadcast for assignment.

Section 8 describes a complete operational digital mobile communications satellite system configuration incorporating the fundamental study assumptions together with the conclusions of the voice digitization and digital modulation tradeoffs. The operational system description includes the system concept, hardware considerations, design parameter selection, and operational procedures for mobile voice communications, low-rate data operation, and demand assignment. The DAMA network concept, hardware, and associated DAMA call sequence procedures are described in depth. Also covered are the considerations involved with use of voice-activated carriers and receiver carrier acquisition. In addition to the operational description, a cost model of the mobile terminals is developed, exploiting the application of LSI technology, in sufficient detail to permit realistic cost estimates.

The field verification experiment plan designed to demonstrate the feasibility of a digital mobile voice communication satellite system using the selected digital voice techniques is presented in Section 9. The baseline for this plan uses off-the-shelf conventional rack-mounted mobile equipment wherever possible. In addition to the technical planning of the experiment, a cost estimate is developed for carrying out the field demonstration.

A glossary of terms and notation is included at the rear of the report as a convenience to the reader.

2.0 EXECUTIVE SUMMARY

This study has demonstrated the viability of using digital voice modulation techniques for mobile communications within the framework of a FDMA-SCPC satellite system serving a large population of land mobile users. The attendant considerations, tradeoffs, studies, developments, and estimates were performed with the key directive of providing a low-cost, low-burden mobile terminal voice communication system suitable for public agency services using relatively untrained operators. The objectives and goals set forth for this study have been met using digital techniques within the constraints of the study assumptions and issues presented in Section 1. This position alone sets the course followed by the digital mobile communications via satellite (DMCVS) study, as detailed in the remainder of this report.

Digital voice modulation techniques have been proven to offer advantageous application to mobile terminal voice communications via satellite in terms of a number of factors. These include system performance, graceful degradation, simplicity of terminal operation, cost reduction implications, low development risk, system flexibility, and inherent compatibility of digital voice, low-rate data, and demand assignment multiple access (DAMA)

The particular combination of voice digitization and digital modulation techniques recommended here provides acceptable voice communications which, at the nominal design point, approaches toll quality speech. In fact, the key to achieving a viable digital mobile communication system is a high tolerance to errors and especially a very graceful degradation characteristic as power is reduced. At low power levels, typical of low-cost mobile communication systems, the digital approach can provide usable voice communication, albeit of degraded intelligibility and quality; whereas a comparable analog FM system, operating at the same power level, fails due to a sharper degradation characteristic. For the DMCVS application these considerations lead to a strong preference for digital voice modulation over analog FM modulation.

A summary of the results of carrying out the DMCVS study is highlighted in capsule form in the discussions that follow.

2.1 Voice Encoding and Digital Modulation

Culmination of the investigation and tradeoff to select the best combination of voice digitization and digital modulation technique has led to the following preferences:

- Continuously variable slope delta modulation (CVSD) for suitable voice encoding.
- Biphase phase shift keying (BPSK) for power efficient digital modulation transmission.
- Operation at a data rate of 16 kbps with a bit error rate of 10^{-3} as a compromise between speech intelligibility/quality and transmission efficiency.
- Quadriphase PSK, known as QPSK, also is acceptable for use as a means of reducing the required channel bandwidth at the expense of higher implementation margin and increased terminal cost/complexity.

Selection of CVSD for voice digitization operating at 16 kbps with BPSK digital modulation leads to a high quality, efficient mobile voice system permitting simplicity and low-cost. At a nominal design operating point with a bit error rate (BER) of 10^{-3}, these techniques provide highly intelligible speech with a carrier power to noise power spectral density ratio (C/kT) of 48.8 dB-Hz to which implementation loss must be added. However, the combination of CVSD and BPSK techniques has an intelligibility/quality limit at a higher BER of about 10^{-1}, which requires 7.7 dB less C/kT or 41.1 dB-Hz (ideal). Hence, these digital techniques inherently provide a sizable performance margin below the design operating point. A high-quality, high cost hardware design has a small implementation loss of about 1 dB, whereas 2.5 dB is typical of low-cost designs. Therefore, depending on the implementation, the recommended digital voice modulation system operates at the design point with 49.8 to 51.3 dB-Hz C/kT and can continue to operate down to the end of the usable range with 42.1 to 43.6 dB-Hz C/kT.

The considerations and decisions that were made in arriving at the above digital techniques selection were carried out within the context of a land mobile voice terminal using satellite repeater communication channels. The primary issues that served to direct the selection process are listed below:

- Fixed mobile antennas (with no pointing) require omnidirectional, hemispherical coverage and consequentially provide relatively low gain.
- In a low-cost mobile environment the system is severely down-link power limited.
- Power efficien[illegible]tal modulation schemes, such as BPSK and QPSK, are the only af[illegible]able types.
- Intelligibility and quality ratings are used here for comparing analog and digital speech systems, since signal-to-noise ratio (SNR) is inappropriate for mobile communication systems operating at low power levels.
- CVSD voice digitization in conjunction with PSK digital modulation offers a very graceful performance degradation as power decreases below normal operating levels to provide a comfortable margin.
- CVSD is the most commonly accepted medium-rate voice encoding technique and available at very low cost.

In addition to the primary issues, several ancillary issues that contributed to this selection process include the following points:

- Use of forward error correction (FEC) is not very effective because of the relatively high error rates associated with a mobile voice communication system.
- Other considerations associated with FEC are the high cost and attendant complications arising with the use of voice-carrier activation (VOX).
- VOX operation can provide significant improvement in spacecraft power utilization and applies mainly to full duplex operation.
- Also VOX operation creates certain receiver reacquisition problems since the carrier is being rapidly turned off and on randomly.
- Backoff of the satellite TWTA is necessary in a SCPC system to hold generated intermodulation crossproducts to acceptable levels at the expense of wasting some available signal power.
- At the present time vocoders operating at 2.4 kbps appear prohibitively expensive, but as technology and the marketplace improve, channel vocoders or linear predictive coder (LPC) vocoders may offer a future alternative to CVSD.

2.2 Traffic Model

Traffic sizing to determine the required maximum number of satellite channels to provide a certain level of service is based upon a specific traffic model. Since no particular traffic information was specified for this study, a traffic model was developed by extrapolation of existing data and by utilizing recommended channel allocations for the mobile requirements of a specific public agency Fire, Emergency Medical, and general Emergency Services (FEMES) system. The developed characteristics of a FEMES network serving the needs of the nationwide CONUS* region are summarized as follows:

• Equivalent number of FEMES Units		44,980	(CONUS)
• Calls per mobile unit	(Peak)	14.8	Calls/hour
	(Average)	4.24	Calls/hour
• Average length of calls		11.47	Seconds
• Total number of calls		184.9	Calls/sec
• Peak traffic intensity		2120.8	Erlangs
• Number of required satellite Channels (Half Duplex)		1917	10% Blocking
		2148	1% Blocking

In designing a communications network for a FEMES application it is crucial to use the peak traffic requirements rather than average traffic requirements. This preference is based upon the critical nature of responding quickly to an emergency call and on its associated specialized calling procedues (such as multiple calls per dispatch). Also, in line with this consideration, a low blocking probability of, say, 1% is preferred to 10% blocking. Therefore, to serve the entire CONUS region a FEMES network requires 2148 channels at half duplex operation for the peak load with 99% availability.

In summary, we recommend adoption of at least 2200 satellite channels for half duplex operation and 4400 channels for full duplex operation to provide a CONUS-wide FEMES network with 1% blocking and including provision for extra channels for the DAMA network control function. Clearly, the provision of such

*Continental United States

a large number of satellite channels for full CONUS coverage forces the employment of a multiple beam antenna (MBA) satellite configuration involving regional spot beams to provide sufficient satellite effective isotropically radiated power (EIRP).

Although the sizing recommendation made here is based upon a specific derived traffic model having a special application, the sizing results have a much broader range of applicability involving variable traffic statistics and grades of service. The only important factors are the total traffic intensity offered to the network and the grade of service provided. In fact, the recommended satellite sizing has the capability to serve a wide variety of traffic models with an excellent grade of service and to continue to serve expanded levels of traffic intensity with lower grades of service.

2.3 Operational System Description

After performing the study and selection of voice encoding/modulation techniques, carrying out a link analysis, and establishing demand assignment requirements; an operational digital mobile communications satellite system configuration was developed taking into account the following factors:

- Incorporation of the basic study assumptions with the results of voice digitization and digital modulation techniques tradeoffs.
- Development of system concept and operational procedures.
- Consideration of hardware impact and selection of system design parameters.
- Provision of capability for mobile voice communications, low-rate data operation, and demand assignment multiple access.
- Application of LSI technology for cost minimization.
- Estimation of mobile terminal costs for quantities ranging from 10 to 10,000 units.

The results of combining these key ingredients has led to the operational system concept illustrated in Figure 2.3-1. Several highlights of the operational configuration are the following:

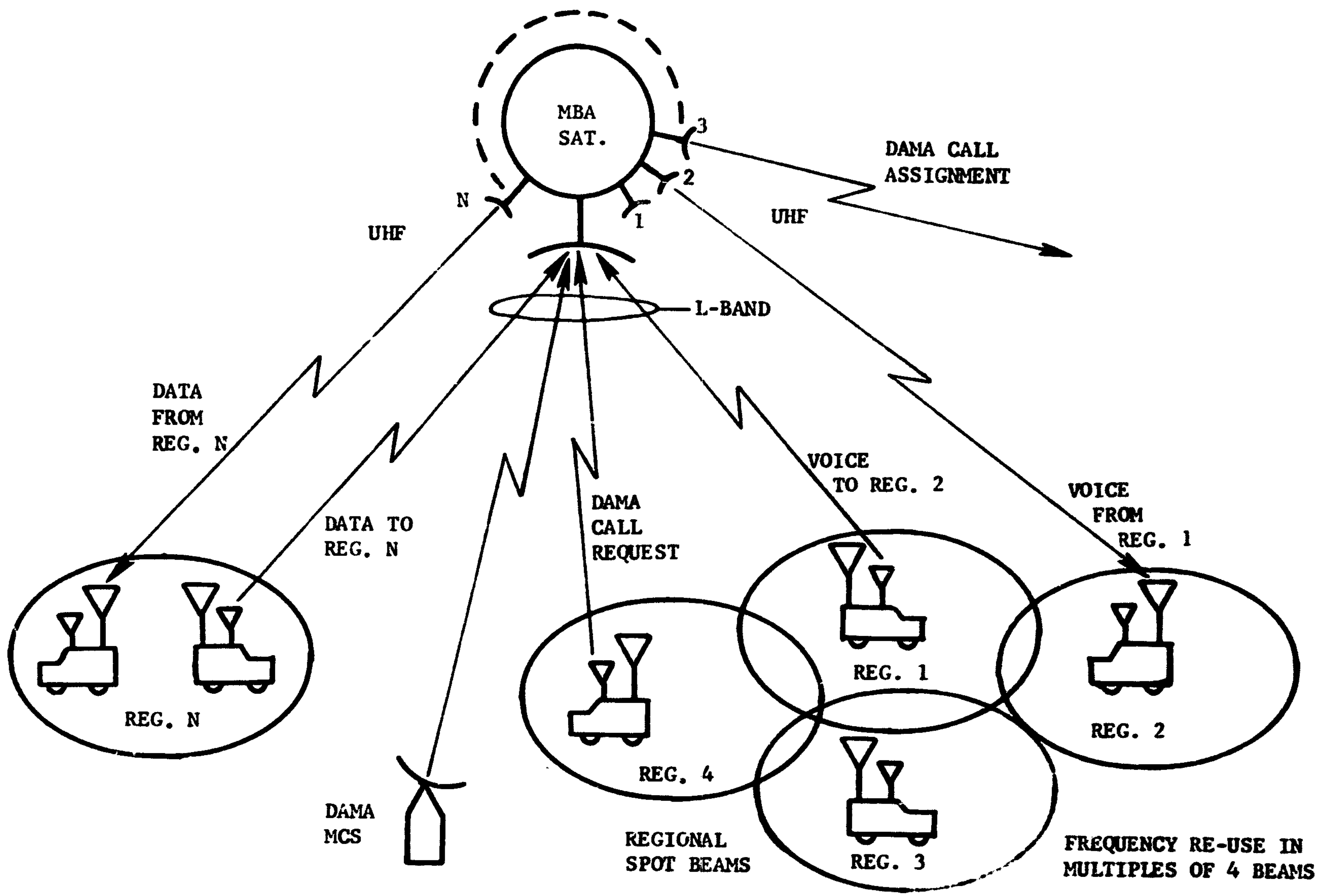

Figure 2.0-1. Conceptualized Operational System

- Multiple beam transmit antenna satellite configuration with CONUS coverage using regional spot beams.
- Common global receive satellite antenna.
- Dual frequency bands; L-band up-link and UHF down-link.
- Up-link, contiguous FDMA; down-link, re-use FDMA in multiples of 4 beams.
- SCPC digital voice, low-rate data, and DAMA.
- Master control station for DAMA operation and control.
- Capacity to support 1125 FEMES mobile units per spot beam with full duplex operation and 99% availability.
- Graceful degradation of performance below the design point.

A comprehensive summary of the operational system parameters and specifications is presented in capsule form in Table 2.3-1. The operational system concept involves a multibeam antenna satellite with a large number of down-link spot beams operating in the mobile UHF band and a single, common global up-link receive antenna operating in an unspecified frequency allocation at L-band. The entire CONUS area is divided up into a number of regional service areas, possible as many as 80. Some portion of two or more adjacent regional areas may lie in a common overlap area, especially along the crowded eastern seaboard. Each regional area in CONUS is served by a separate satellite spot beam in order to provide sufficient satellite EIRP. Each mobile ground terminal communicates directly with any other mobile ground terminal by means of an appropriate SCPC-FDMA channel assignment.

The broad FDMA up-link frequency band is at L-band, where more bandwidth is available, and the narrower down-link frequency band is in the 806 to 890 MHz UHF mobile band. The satellite filters and translates the N up-link frequency bands to N separate down-link spot beams that are uniquely dedicated to the N service regions. This use of dual frequency bands for transmit and receive is forced by the lack of sufficient frequency space at UHF to accommodate the large number of channels anticipated for full CONUS coverage.

Each service region is assigned a specific FDMA frequency band on the up-link at L-band. The individual frequency bands accommodate 56 channels, two of which are reserved for DAMA control. Within each frequency band the 56 channels are spaced by 45 kHz, a channel spacing commonly used for SCPC systems.

Table 2.0-1. Operational System Parameters and Specifications

ITEM	SPECIFICATION	COMMENT
FREQUENCY PLAN	FDMA-SCPC	DUAL BANDS
• UP-LINK	L-BAND (UNDEFINED)	CONTIGUOUS FDMA
• DOWN-LINK	UHF MOBILE	RE-USE FDMA
SATELLITE CONFIGURATION	MULTIBEAM ANTENNA	FREQUENCY TRANSLATES IN MULTIPLES OF 4 BEAMS
• UP-LINK ANTENNA	GLOBAL RECEIVE	CONUS COVERAGE
• DOWN-LINK ANTENNA	SPOT BEAM TRANSMIT	REGIONAL COVERAGE
NUMBER OF CHANNELS (FULL DUPLEX)	56/FREQUENCY BAND, UP TO 80 BANDS	2 CHANNELS/BAND RESERVED FOR DAMA
BANDWIDTH	45 kHz/CHANNEL 3.02 MHz/BAND	SPECTRAL SHAPING (?) 0.5 MHz GUARD BAND
VOICE ENCODING	16 kbps CVSD	7.7 dB INTELLIGIBILITY MARGIN BELOW DESIGN POINT
DIGITAL MODULATION	BIPHASE PSK	VOICE-ACTIVATED CARRIERS
CARRIER-TO-NOISE DENSITY	51.3 dB-Hz	10^{-3} BER, 2.5 dB LOSS
LOW-RATE DATA	VARIABLE TO 16 kbps	MICROPROCESSOR CONTROLLED
DAMA NETWORK	CENTRALIZED	MICROPROCESSOR CONTROLLED
• SIGNALLING	16 kbps	SHARED MODEM W VOICE/DATA
• CALL REQUEST	1 CHANNEL/BAND	RANDOM ACCESS, UNSLOTTED
• MCS ASSIGNMENT	1 CHANNEL/BAND	COMMON MODE BROADCAST
MOBILE TERMINAL		
• RF POWER	40 WATTS	
• RECEIVER G/T	-26 dB/^{o}K	EDGE OF COVERAGE
• ANTENNAS	OMNIDIRECTIONAL-HEMISPHERIC BENT MONOPOLE-TURNSTILE	UHF RECEIVE, L-BAND TRANSMIT
SPACECRAFT EIRP	50 dBW/SPOT BEAM	3 dB BACKOFF, 4.5 dB VOX
SPACECRAFT G/T	3 dB/^{o}K	
MOBILE TERMINAL TRAFFIC	1125/SPOT BEAM	FULL DUPLEX, 1% BLOCKING

The UHF down-link frequency bands are unique only to the adjacent service regions. If service Region 1 is bounded by Regions 2, 3, and 4, then Region 1 requires a unique frequency band assignment apart from Regions 2, 3, and 4. However, the frequency band assigned to Region 1 may be used over again for any other region not adjacent to Region 1 nor adjacent to another region that used the same frequency band.

To be viable this type of frequency plan requires the coordination and supervision of a demand assignment multiple access (DAMA) system for random access call requests and channel assignments. The DAMA configuration is a centralized system using a single earth station for network control which is referred to as a Master Control Station (MCS). The MCS uses a minicomputer system to control all stations, assign frequencies, and maintain status of all terminals and channels in every spot beam. Each user terminal requires a microprocessor for station control. To minimize the total network cost, each mobile terminal is provided with only one modem which is used for both signalling with the MCS and for communicating with other mobile terminals (i.e., shared between the DAMA mode and the voice mode).

Certain desirable characteristics of the DAMA network are listed below:

- Transparent to users
- Minimize post dialing delay
- Minimize number of signalling channels
- Minimize overall system cost
- Capable of system clearing when overloaded
- Capable of priority assignment and break-in

One of the key items listed is the operational requirement that the DAMA system be transparent to the user. The DAMA system should process calls in a manner that the user is familiar with; for example, the commercial telephone system. In point of fact, the operational scenerio proposed for the DAMA operation appears to the user as a conventional telephone network and thus meets the objective of placing a low burden on the potential users of the operational system.

Basically the operational system described here employs biphase PSK digital modulation in conjunction with continuously variable slope delta modulation (CVSD) voice digitization operating at a 16 kbps data rate. This combination of modulation and voice encoding techniques offers good quality system performance featuring graceful degradation together with low terminal complexity and a corresponding low cost. Options are provided for handling data services at any rate up to 16 kbps plus the shared modem concept for demand assignment multiple access for large terminal populations.

A unique feature of the BPSK data demodulator described here is the proposal to employ a virtually all-digital implementation. This design is a digitally-implemented version of the classical analog Costas PLL and offers the advantage of being relatively inexpensive to implement compared to conventional analog circuitry. Moreover, this all-digital approach achieves measured performance to within only 1.2 dB of theory and to be operable at low received signal levels.

Another factor contributing to low-cost modem designs is the adoption of a large overall modem implementation loss margin of 2.5 dB, that includes the implementation loss of the all-digital demodulator.

QPSK modulation is an alternative modulation method that could be used here to achieve a 50% savings in required channel bandwidth. However, this advantage is offset by greater modem complexity that translates into higher cost and a larger implementation loss. This condition is due to an inherent sensitivity to channel impairments and imperfections that is aggravated by the use of VOX operation (the requirements for the carrier reconstruction loop become excessive). Therefore, for this application BPSK is preferred over QPSK in the interest of low-cost, simplicity, minimal implementation loss, and relaxed VOX requirements.

2.4 Terminal Cost Estimate

In addition to the operational description, a cost model of the mobile terminals has been developed exploiting the application of LSI technology wherever possible for quantities of 10 to 10,000 terminal units. The design description of the mobile terminals was developed in sufficient detail to a level that permits realistic cost estimates of the operational system. Several key factors identified as relevant to determining terminal costs include:

- Detailed terminal hardware configurations differ significantly at each quantity level for cost minimization.
- LSI and hybrid technologies require large initial investment.
- Large quantity production runs necessary to achieve low-cost per unit for LSI and hybrid circuitry.
- Large initial investment required for the DAMA Master Control station.
- DAMA operation economically unfeasible for lower quantities.

A cost summary of mobile terminal cost estimates without provision for low-speed data or DAMA options appear below:

Number of Units	Cost per Unit (Voice only)
10	$64,380
100	$20,946
1,000	$ 5,756
10,000	$ 3,947

These cost estimates include both recurring and non-recurring costs for mobile terminals with voice capability only. For large terminal populations, such as 10,000, the additional cost of providing both low-speed data and DAMA capability is expected to be only $610 per terminal for a net terminal cost of $4,557. (DAMA contributes about 75% of this additional cost.)

2.5 Field Demonstration Experiment Plan

The technical objectives and goals involved in planning and carrying out the field verification experiment are highlighted below:

- Plan field verification experiment.
- Demonstrate feasibility of digital mobile voice communication via satellite.
- Utilize selected voice digitization and digital modulation techniques.
- Conduct laboratory and field tests of performance, techniques, parameters, and reception quality.
- Verify and compare performance characteristics of voice digitization methods and implementations.
- Evaluate and document experimental results and conclusions.

In addition to the technical planning aspects of the demonstration, a costing exercise was undertaken to estimate the cost of performing the field experiment.

For the field demonstration and verification tests, the ATS-6 satellite would be utilized in the 1535 to 1565 MHz receive band and in the 1630 to 1670 MHz transmit band, making use of the L-band pencil beam satellite antenna.

The experiment involves two mobile earth terminals, vehicularly mounted, and primarily designed to verify and evaluate a voice-operated, digital mobile communication system. Testing will be performed for a range of system parameter values with a BPSK modem operating at 16 kbps and utilizing CVSD for voice encoding. Narrowband FM modulation is included in order to compare the digital approach to the conventional analog FM-SCPC scheme.

A baseline taken for this experiment calls for the use of off-the-shelf, conventional rack-mounted mobile equipment wherever possible to minimize costs. This baseline was followed throughout to the point where only two items, the L-band mobile antenna and the digital baseband BPSK modem, require minor development work. However, this development consists mainly of retrofitting or modifying existing hardware to meet the specifications of the experiment. The main reason for this lack of available suitable modems is that the use of VOX imposes a stringent requirement of quick acquisition and reacquisition time, 2 to 4 msec, at low E_b/N_o values ($<$ 3 dB). Aside from these two items, all of the experimental equipment is standard, utilizing present-day solid-state technology as much as possible. In fact, the resulting equipment configuration was designed around these available components for system parameters such as gains, noise figure, power levels, etc.

2.6 Experiment Plan/Test Paramaeters

The proposed experiment plan is divided into three distinct phases consisting of:

- Phase 1 - System design, procurement, and hardware implementation of two mobile L-band earth stations.
- Phase 2 - In-lab checkout and pre-testing to establish a performance reference preparatory for field testing.
- Phase 3 - Field testing, experimentation, and evaluation of the data obtained from the field tests.

During the laboratory and field testing, several system parameters and quality factors would be evaluated including:

- Link margins, implementation margins, and antenna performance.
- Bit error rate, E_b/N_o, and degradation effects at low C/kT ratios.
- Earth station system Figure of Merit, G/T, and carrier to noise density ratio, C/kT.
- Informal speech intelligibility assessment and comparative reception quality rating for both the digital and the analog FM methods (input and output signal-to-noise ratios for FM).
- Voice-activated carrier operation (acquisition times, lock-up threshold, etc.).

The above parameters are to be measured prior to field testing either individually or on a loop basis, as appropriate, with conditions in the lab closely simulating conditions to be encountered in the field for realistic evaluation and to establish a reference point for the field tests.

The results of the field test data would be reviewed, evaluated, and compiled into a final report describing the equipment, test procedures, and recommendations.

Completion of the full field demonstration experiment is expected to require 8 to 11 months, depending upon the scope of the actual field testing.

The physical size of the experimental equipment is expected to be about the equivalent of two standard racks, 16" wide by 8" high by 16" deep, one each for

the RF section and for the baseband section. The L-band antenna is approximately 8" in diameter by 2" high mounted on a ground plane of about 18" by 24". This hardware will weigh approximately 40 to 50 lbs. for the RF section and 20 to 30 lbs. each for both the baseband section and the antenna.

2.7 Experiment Cost Summary

A complete cost baseline and a detailed cost breakdown are described in Section 9.7. The final cost estimates including both hardware and labor at 1978 prices and wages are summarized as follows:

Phase 1 - Design and hardware implementation of two mobile earth stations	$118,100
Phase 2 - Laboratory checkout and verification tests	$ 12,600
Phase 3 - Field verification experiments	$ 38,800
Miscellaneous	$ 500
TOTAL	$170,000

These cost estimates are based on two complete baseband modems loaned to NASA free-of-charge, other than the recurring cost, for 3 to 4 months, which is the estimated length of time to complete Phases 2 and 3 of the experiment. However, if NASA exercised the option to purchase the modems, the additional cost would be about $14,000 for a total of $184,000. No cost has been assigned for the use of the satellite other than a small cost for satellite scheduling consisting mainly of an administrative function. Also not included are the costs of general administration and overhead normally incurred in carrying out a project.

2.8 Recommendations for Further Investigation

In the course of carrying out this study several technical areas and topics have been identified as meriting further study and in-depth review. These items, generally speaking, heavily impact either the mobile system performance or the study conclusions. Recommended areas for further detailed investigation are listed below:

- The mobile terminal antenna is a critical element limiting the performance of a mobile communications satellite system. Higher gains are needed, such as provided by directional antennas, and a potential solution is a small electronically-steered array.
- Spacecraft antennas, particularly large multibeam UHF or L-band antennas need to be developed along with higher spacecraft EIRP.
- Digital modems capable of operation at low C/kT values and fast acquisition/reacquisition are necessary to achieve the potential large performance margins offered by digital voice modulation.
- Higher frequency bands (L-band to C-band) should be investigated as a means of providing sufficient bandwidth to accommodate the requirements of a nationwide public service satellite communication system.
- Further testing and performance comparison is needed to establish a firm basis for verification and evaluation of subjective intelligibility and quality ratings of several different voice digitization techniques operating at variable power levels and data rates.
- Since suitable data presently doesn't exist, comprehensive traffic studies aimed at meeting the diverse needs of various government and public service agencies should be performed to establish the sizing requirements of mobile terminal satellite systems serving specific purposes.

3.0 TRAFFIC ANALYSIS

Although there was no specific requirement for traffic analysis included in the study objectives, nor was a traffic model involved in the basic study assumptions, it became evident at the outset that some form of traffic sizing (system usage) is a necessary part of the study plan. This item is needed because the sizing of a communication satellite system requires some measure of the expected traffic flow on which to base the determination of the required maximum number of satellite channels to provide a desired level of service. In fact, to our knowledge no such data exists as to the probable traffic requirements of a land mobile voice communication system, nor its specific application. Experience has shown that when traffic studies are available and future traffic projections made, actual system usage quickly outstrips the planned traffic activity (e.g. the INTELSAT system). Since this type of information was not readily available, an attempt is made here to speculate as to what the traffic requirements might be for the digital mobile voice communication satellite system. While the postulated traffic model is aimed at a special application, it is shown that the results have a much broader range of applicability involving variable traffic statistics and grades of service.

The basis of this speculation is a recently completed study (Ref. 3-1) of mobile emergency services communication requirements for Santa Clara county, California. This project examined the joint city-county law enforcement, fire, and emergency medical services mobile radio communication requirements for the county during the period of 1974 and made projections for 1990. The task at hand here is to extrapolate these results from the county level to the entire CONUS (continental U.S.) region. While it is true that the sample data size is small, hopefully it is somewhat representative of the nation-wide requirements since Santa Clara county includes a rather diverse mixture of an urban industrial-commercial complex as well as suburban and rural areas.

The focus of this analysis is on the mobile requirements of fire, emergency medical, and general emergency services (FEMES) rather than on law enforcement. Unfortunately, the referenced sample study devoted the majority of its attention to the law enforcement aspects, since this comprised the bulk of the traffic. Thus, the available data is somewhat slanted toward law enforcement and needs to be interpreted for the FEMES application, which is of greater concern here.

A summary of Santa Clara county mobile radio law enforcement traffic usage is presented in Table 3.0-1. The data was tabulated for a mobile population of 771 active units and at the peak of the busy hour made 14.8 calls per hour per unit as compared to an overall average of 4.24 calls per hour per unit. The average length per call was only 11.47 seconds reflecting the somewhat specialized nature of police dispatch procedures.

Table 3.0-1. Santa Clara County Law Enforcement Mobile Radio Usage (1974)

• TOTAL NUMBER OF MOBILE UNITS		771	
• TOTAL NUMBER OF CALLS	PEAK	11,411	CALLS/HOUR
	AVERAGE	3,269	CALLS/HOUR
• CALLS PER MOBILE UNIT	PEAK	14.8	CALLS/HOUR
	AVERAGE	4.24	CALLS/HOUR
• AVERAGE LENGTH PER CALL		11.47	SECONDS

To translate this data to traffic requirements associated with FEMES is not an easy task because of the difference in the type of required service. Observations indicate that both fire and emergency medical calling procedures involve more calls per dispatch than are generally required for law enforcement and that the calls may tend to be of longer duration. Also, the delay time in responding to a FEMES request is more critical due to the emergency nature of the call. In fact the peak to average usage ratios can be on the order of 100 to 1 influencing the sizing to be done on the basis of the peak traffic. For this reason, the number of mobile radio channels allocated to fire, medical, and emergency services is disproportionally high compared to the relative traffic volume.

Table 3.0-2 shows the percentage distribution of actual Santa Clara county public service dispatches by type of service and the associated channel allocation recommendations determined by the traffic study (Ref. 3-1).

Table 3.0-2. Santa Clara County Public Service Traffic Distribution and Channel Allocations

	DISTRIBUTION OF S.C. COUNTY PUBLIC SERVICE DISPATCHES	RECOMMENDED CHANNEL ALLOCATIONS
• LAW ENFORCEMENT	88%	14 CHANNELS
• FIRE	9%	5 CHANNELS
• MEDICAL AND EMERGENCY SERVICES	3%	2 CHANNELS
(COMBINED FEMES)	(12%)	(7 CHANNELS)

Obviously the percentages of actual radio dispatches are not indicative of the recommended channel allocations, but do reflect the critical nature of the FEMES traffic. It is apparent that in order to provide adequately for this type of service, a better sizing practice is to follow the recommendations of the study and assign sizing ratios of 67% (14 out of 21) for law enforcement and 33% (7 out of 21) for FEMES. The effect of using this relative sizing is to consider that there are more FEMES mobile units than actually exist having the same traffic statistics as law enforcement. Using this approach, the equivalent number of FEMES mobile units is 771/3 = 257 units for Santa Clara county.

The next plausible step in developing a representative traffic model for CONUS is to extrapolate the data on the basis of population using 1974 estimates. Table 3.0-3 presents the results of this extrapolation from S.C. county to all of the state of California and to the entire country.

Table 3.0-3. Extrapolation of Traffic Call Data For FEMES

	S.C. COUNTY	CALIFORNIA	CONUS
• POPULATION (1974)	1.2 M	21 M	210 M
• EQUIVALENT NUMBER OF FEMES UNITS	257	4,498	44,980

As indicated, this traffic modelling requires accommodating nearly 4,500 equivalent FEMES mobile units for the state of California and about 45,000 units for CONUS.

In a communication network the traffic intensity or grade of service is measured in terms of erlangs, defined as the product of the traffic arrival rate and the average call duration. An erlang is given in units of one call-second per second. For example, here the peak traffic rate per mobile unit is 14.8 calls per hour or 0.00411 calls per second and using an average length per call of 11.47 seconds, each mobile unit transmits a peak traffic intensity of 0.00411 x 11.47 = 0.0472 erlangs (257 mobile units comprise 0.0472 x 257 = 12.12 erlangs). Network sizing is based on the total traffic intensity offered to the network.

Another concept needed here is the Erlang B equation given by

$$P_B = \frac{t^N/N!}{\sum_{n=o}^{N} t^n/n!}$$

where

P_B = Probability of blocking

t = Traffic intensity in erlangs

N = Number of channels (or trunks)

This formula computes the probability of blocking (i.e., of denying a user access because the network is momentarily loaded to capacity) given a level of traffic intensity and a number of channels to serve the users. Together with the traffic model this equation permits determination of the sizing of the mobile network.

Note that while the traffic statistics per mobile terminal remain fixed at rates of 14.8 calls/hour peak, 4.24 calls/hour average, and average duration of 11.47 seconds/call, the recommended scaling from law enforcement to equivalent FEMES units reduces the total traffic offered to the network by a factor of 3. For example, for FFMES the total number of calls per hour at the peak activity is estimated to be 11411/3 = 3804 or 1.057 calls per second. Table 3.0-4 shows the total traffic rate, the offered traffic intensity, and the required number of channels to provide either 10% or 1% blocking for a FEMES network serving Santa Clara county, the state of California, or the entire CONUS.

Table 3.0-4. FEMES Traffic Model and Channel Requirements

	S.C. COUNTY		CALIFORNIA		CONUS	
	AVG.	PEAK	AVG.	PEAK	AVG.	PEAK
• TOTAL NUMBER OF CALLS/SEC.	0.303	1.057	5.299	18.49	52.99	184.9
• TRAFFIC INTENSITY (ERLANGS)	3.48	12.12	60.78	212.08	607.8	2120.8
• NUMBER OF REQUIRED CHANNELS						
10% BLOCKING	6	15	60	198	555	1917
1% BLOCKING	9	20	75	233	635	2148

Data is presented in Table 3.0-4. for both average traffic rates and for peak traffic rates; however, in designing a communications network for a FEMES application it is best to use the peak traffic requirements. This preference is based upon the critical nature of responding quickly to an emergency call and on its associated peculiar calling procedures. Also, in line with this consideration, a low blocking probability of, say, 1% is preferred to 10% blocking. Therefore, to serve the entire CONUS region a FEMES network requires 2148 channels at half duplex operation for the peak load with 99% availability.

In summary, we recommend adoption of at least 2200 channels for half duplex operation and 4400 channels for full duplex operation to provide a CONUS-wide FEMES network with 1% blocking and including provision for extra channels for DAMA network control.

Furthermore, it becomes evident at this point that the large number of satellite channels required for full CONUS coverage forces the employment of a multibeam antenna satellite configuration to provide sufficient link EIRP. This contention is supported by the considerations presented in Sections 5 and 6.

The recommendation made here for sizing a CONUS-wide satellite system serving mobile terminals is based upon a specific derived traffic model suitable for a FEMES network in which the average call length is rather short (11.47 sec.) and the peak call activity is rather high (184.9 calls/sec.). However, this sizing recommendation applies to a broader range of traffic models. In fact,

the only important elements are the total traffic intensity (2120.8 Erlangs) offered to the network and the grade of service provided (1% blocking). The traffic intensity is, as noted above, the product of the calling rate and the average call length. As long as this product remains constant, the sizing requirement is unchanged. For example, 2148 channels also provide for 1% blocking (half duplex) with intermediate call durations of 114.7 sec. arriving at 18.5 calls/sec., or long call durations of 1147 sec. at a rate of 1.85 calls/sec., etc.* If the total traffic intensity increases, the system continues to operate but provides a lower grade of service (less availability). For example, for 2148 channels a 12.5% increase in traffic intensity results in 10% blocking, a 25% increase yields 19% blocking, and a 100% increase (i.e., doubling the traffic intensity) yields 49% blocking. Thus, it is apparent that the recommended sizing can serve a variety of traffic models with an excellent grade of service and can continue to serve expanded levels of traffic intensity with lower grades of service.

*Long call durations are more typical of low rate data transfer usage than voice communications.

4.0 VOICE DIGITIZATION TECHNIQUES

The technology of single-channel-per carrier (SCPC) satellite communication systems has been developed extensively (Refs. 4-1,2). This study extends this technology to digital mobile voice communications. Special characteristics that are expected for mobile radio links include multipath, fading, variable signal strengths, and sub-optimal mobile antenna positioning (fixed, hemispherical coverage). Several different voice digitization and digital modulation techniques are compared with analog FM modulation, for a wide range of signal-to-noise ratio (SNR), bandwidth, and voice quality. Although the numerical ranges for SNR and bandwidth can be relatively easily specified, voice quality is a much more elusive and complicated property not easily parameterized. The various measures of voice quality that are used commonly in the literature are discussed in a following section. In summary, we can say that SNR can be a fairly good descriptor of toll quality speech (greater than 30 dB input SNR), but for low quality communication that is marginally usable in air traffic or police applications, many different characterizations are possible. For example, 0-10 dB input SNR, reduced input bandwidth, and modified rhyme test (MRT) intelligibility scores of 75% (Ref. 4-3) are just a few. These characterizations are very dependent on the type of voice communication system being described, and hence system comparisons are difficult. Since SNR is not a good descriptor for mobile voice communication systems, we must look to other performance measures in terms of intelligibility and quality versus available link power ratios.

Previous studies that have considered small mobile ground stations (Ref. 4-4) have concluded that analog FM modulation is superior to digital modulation. This comparison was made for only one value of carrier power to noise power density ratio (C/kT), and the margin of difference was only 2.5 dB, which is less than many of the allowance factors commonly introduced for noise effects not adequately described by SNR (such as the FM impulse noise). In this section that conclusion is examined in a more thorough fashion over a wider range of voice quality measures and link conditions.

This study included a literature review of existing SNR performance studies for digital voice techniques and summarized those results which are comparable. The summary indicates a vast diversity of SNR definitions and performance measures and points up the inability to make direct comparisons due to that diversity. For this reason, and in many cases because of the inappropriateness of SNR measures for performance evaluation, the primary basis employed here for comparison of analog and digital voice communication systems is intelligibility and quality ratings. These results show that for mobile voice communications digital methods provide superior performance at lower power levels than conventional analog methods. This conclusion is unique since the general concensus in the literature is that digital modulation is slightly inferior to analog (Ref.4-5).

Thus, digital coding of voice signals offers several advantages including:

- Graceful degradation for increasing channel errors and decreasing power levels
- Compatible with existing satellite transponders and demand assignment procedures
- Compatible with digital land line systems
- Offers efficient signal regeneration
- Requires less stringent linear transponder
- Allows easy encryption
- Permits a combination of switching and transmission functions
- Offers a uniform format for different signals
- Uses low-cost devices for implementation
- Allows a tradeoff of bandwidth and power for a specific channel error rate over a larger range of link power ratios

Thus, the current state-of-the-art supports the selection of digital voice modulation with complementary demand assignment and low-rate data transmission techniques, for land mobile satellite communication.

4.1 Voice Encoding Methods

Various methods of digital waveform encoding have been developed for speech signals and were surveyed by Jayant (Ref. 4-6). There are also many encoding schemes that use speech production models to significantly lower the encoded data rate (vocoders), but at the present time their cost is still too high to offset the advantage of the lower bandwidth requirements for this particular mobile application. However they will be included here for comparison purposes. Furthermore, all of the various voice digitization techniques perform better in the adaptive form, and thus nonadaptive techniques will not be considered (with the exception of PCM, where companding is equivalent to adapting the quantizer step-size). While the encoding can be optimized for specific signals, encoding quasi-stationary speech signals benefits from adaptive techniques where the quantizer step-size is matched to the dynamic signal characteristics, or where the predictor estimates the short-term signal properties of the input speech. Syllabic companding (time constants on the order of 5-10 msec) is better for mobile systems, especially when channel errors occur.

The following is a brief overview of the voice digitization techniques we will consider:

a. Pulse code modulation - PCM is the standard digital voice technique, developed for the phone system and used in the INTELSAT SPADE system, which incorporates A-law companding, 7-bit coding, and a sample rate of 8000 samples/sec. The main disadvantage is high data rate (56 kbps or 64 kbps with PCM framing). Ways to reduce the data rate include coding with a lower number of bits (4-6) plus adding dither noise and nonuniform quantization (which does not provide much improvement over companding). Equivalent noise levels have been worked out for quantization, saturation, and channel bit errors (Ref. 4-7).

b. Adaptive Differential Pulse Code Modulation - With ADPCM an advantage is gained in coding the difference between adjacent signal levels rather than the original signal, if there is a high one-step correlation (C1 > 0.5). This advantage can be in the range of 4-10 dB over straight PCM. However, the exact

amount is highly dependent on the talker, speech material, sound type, etc. Thus, one is naturally led to adaptive (predictor or quantizer) DPCM. Also, the effect of channel bit errors should be considered, since they may propagate due to the feedback in differential decoders (Refs. 4-8, 9).

c. Adaptive Delta Modulation - ADM is a general class of adaptive quantization techniques. When the number of levels in a DPCM system is reduced to two (1 bit), the technique is termed delta modulation. The reduced number of quantization levels is compensated for by oversampling above the Nyquist rate. Delta modulators are well suited to the long term average speech spectrum because of an integrator in the feedback loop (6-dB/octave slope). Since short-term speech characteristics are significantly variable, adaptive quantizers yield performance improvements.

d. Continuously Variable Slope Delta Modulation - CVSD (a slight variant is known as DCDM) is a simplification of the ADM schemes. The quantization levels are continuously adapted according to the digital bit stream: Three successive ones or zeros increase the step-size, and "101" or "010" decrease the step-size. The decoder uses the same algorithm. Sample rates two to three times the Nyquist rate generally give good quality speech. The widespread use of CVSD is due to the simplicity of the algorithm, the ease of varying the data rate (by changing the sample rate), and the preservation of speech quality if bit errors, background noise, or lost data occur. Also, the bulk of waveform speech coding testing has been done on CVSD-encoded speech.

e. Adaptive Predictive Coding - APC is a slightly more complicated (than DM) technique (Ref. 4-10) that shows some promise of improving voice quality for data rates in the 6.5 to 9.6 kbps range. The reduced data rate (and hence smaller bandwidth) will have to be weighted against the increased complexity/cost. Performance in the presence of channel errors must also be evaluated.

4.2 Speech Performance Measures

The speech performance quality of a communication system is a function of many variables. Acceptability of a particular system depends upon the degree to which the needs of the system's user-community are adequately met by the delivered quality of a system. For some applications, barely intelligible speech may be adequate whereas for others, excellent, high fidelity speech may be required. Since the range of potential user needs and the quality of deliverable speech are both very broad, it is difficult to specify a single universal measure to be used in comparison of voice communication systems. One approach, however, is to examine speech intelligibility as a function of the significant satellite system and mobile terminal parameters. In this study, the available down-link power is the limiting parameter, and not bandwidth, so we will use the carrier to noise power density ratio, (C/kT) as the independent parameter. Intelligibility and quality rating results in terms of bit error rate will be converted to C/kT, assuming biphase PSK modulation as a reference.

4.2.1 Assessment of Toll Quality Speech. The Committee Consultat Internationale Telephonie et Telegraphie (CCITT) makes various recommendations concerning criteria to be met by public speech communication networks. For toll quality speech these recommendations specify that channel noise should not exceed 8500 pWOp and that FM channels should not have an impulse count of more than 15 impulses exceeding a level of -18 dBmO per fifteen minutes.

An FM-PLL system could in theory meet such stringent requirements at a bandwidth of 25 kHz and a C/kT of 53.5 dB-Hz (Ref. 4-5). This value of C/kT is about 3 to 4 dB above the FM threshold so as to ensure an impulse count meeting the above CCITT specification. However, the equipment margins required to account for field implementation of low-cost mobile units probably make such a goal unrealizable.

These same requirements could be closely approached by a digital system employing 32 kbps CVSD operating with a bit error rate (BER) of 10^{-4}. Using QPSK modulation, such a system would require about the same bandwidth and C/kT value as the FM system. Again, implementation margins, especially to meet the tight bandwidth, make it unlikely that the CCITT goal could be met within the limits

of a mobile satellite network. However, an interesting point is that very little loss of intelligibility would result at a BER of 10^{-2} requiring 4.1 dB less C/kT, whereas the FM system would fail dramatically.

The cost restrictions of mobile units make it unlikely that complex speech encoding methods such as linear predictive coding (LPC) vocoders can be used at this time. Hence, candidate digital encoding methods are restricted essentially to relatively simple waveform encoding techniques. If such candidate techniques are operated at data rates that provide performance approaching toll quality speech, they will all produce very similar sounding speech. In fact, the reconstructed speech will be of very high intelligibility and excellent overall fidelity, but differences in the degree of "noisiness" of speech and in the kind of noisiness will be perceptible. However, these differences will be small and preferences for one encoding technique or another will differ from user to user. Thus, if toll quality speech service were a required criterion in a mobile satellite network then selection of a particular speech encoding method should probably be based strictly upon considerations such as cost, maintainability, etc. rather than on minimal quality differences. In point of fact, toll quality speech is not only unnecessary but also cost ineffective for an application such as the public service mobile satellite network. Lesser quality techniques are far more appropriate.

4.2.2 Quality and Intelligibility Comparisons for Candidate Systems. Voice digitization technique (VDT) performance is influenced by many factors but a key parameter is the operating speech data rate. In order to make a rough comparison of voice digitization techniques as a function of speech data rate, we consider here that these other factors play a lesser role. In the following, a baseline speech system model is employed with parameters given by:

- Audio BW = 3400 Hz
- Input Speech Level > -16 dBm0
- Input Speech-to-Acoustic Noise Ratio > 15 dB
- BER < 10^{-2}

Using this baseline model, Table 4.2-1 presents an estimate of the kind of speech quality performance to be expected from various VDT operating at different data rates. The subjective quality ratings employed here reflect performance relative to toll quality systems, whereas for mobile applications reduced speech performance may be common. However, toll quality systems provide an appropriate reference, since we are concerned with the subjective comparisons that a naive user would make with familiar standards such as the phone system.

Table 4.2-1. Minimal Data Rates for Various VDT Quality Classes

VDT	EXCELLENT	GOOD	ACCEPTABLE	MARGINAL
PCM †*	88 kbps	63 kbps	35 kbps	14 kbps
DPCM †*	76	56	28	14
LOG PCM *	64	42	28	14
ADPCM *	52	28	14	10 **
ADM		24	12	8 **
CVSD		32	16	10 **

† Included only for comparison, no significant cost or other advantage over LOG PCM.

* Multibit VDT generally require greater implementation costs than single-bit VDT. Also, individual bits are differentially important to quality.

** Effective audio bandwidth probably less than 3400 Hz.

Consequently, PCM systems are listed in Table 4.2-1 since they are the standard voice digitization technique developed for toll quality systems. PCM requires considerable bandwidth for the high data rates and generally provides high quality performance with BER < 10^{-4} However, the required C/kT is on the order of 58.7 dB-Hz, which indicates that PCM methods are not candidates for low

power mobile terminals. Recently, the implementation cost of PCM has been reduced dramatically due to the development of single-chip codecs (such as produced by Intel) and low cost anti-aliasing filters (National's cost ~ $25).

Vocoder techniques are not included in this table because data rate is not the major determinant of vocoder voice quality.

Table 4.2-1 represents a rather coarse summary of quality rating as a function of VDT type and speech encoding data rate. It is the result of incorporating results from several sources such as Figure 4.2-1 from Ref. 4-11. The MRT intelligibility scores for several CVSD and vocoder VDT are shown in Figure 4.2-1 as a function of both input SNR and data rate. Clearly indicated on this figure is the robustness of waveform VDT (such as CVSD) when the input SNR is reduced, whereas the vocoders are severely affected by this reduction. This lack of robustness makes the vocoders barely useable for anything except the very good input SNR conditions (background noise equivalent to a quiet office).

The major factor restricting the usefulness of Table 4.2-1 is the assumption that BER is always less than 10^{-2}. To extend the range of applicability, we have compiled a considerable amount of additional intelligibility and quality rating data for CVSD and vocoder VDT for a wide range of BER. CVSD is emphasized because of the wide-spread acceptance within the Government for voice encoding applications. Also, because of this acceptance, there is much more test data readily available for CVSD and several low-cost LSI implementations do exist (Ref. 4-12). This data was taken directly from Refs. 4-11, 4-3 whenever possible, and interpolated to fill in the gaps. The total compiled data was cross-checked against other data gathered under a DoD government study for validity and consistency. The combined results for diagnostic rhyme test (DRT) intelligibility as a function of C/kT is shown in Figure 4.2-2 for a range of values expected to be encountered in a land mobile terminal satellite application.

Included in Figure 4.2-2 for comparison is intelligibility rating data for a linear predictive coder (LPC) vocoder. Although the speech performance is better than CVSD at low power levels (mainly due to the low data rate) LPC is not considered a viable candidate for the following reasons:

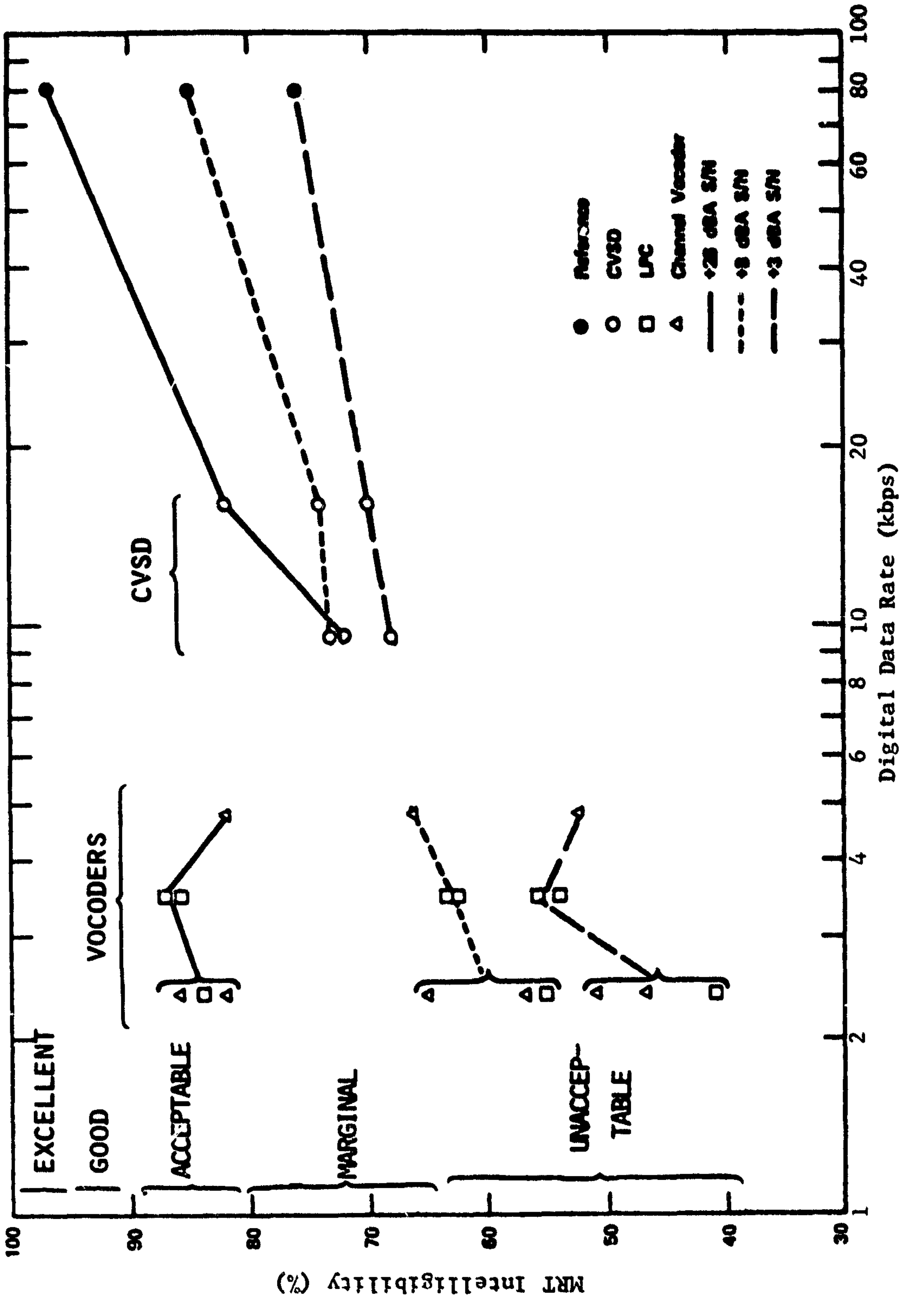

Figure 4.2-1. Intelligibility Comparison of Voice Digitization Techniques

a. Initial speech performance at high power levels is poorer than CVSD operating at 16 and 32 kbps.
b. High input SNR is required (approximately 26 dB or better).
c. Presently (and for the near future) implementation costs are high.

LPC and channel vocoders do hold promise for the future, assuming additional development improves the initial quality and background noise deficiencies and costs decline. LPC vocoders are digital computation oriented approaches based on statistical estimation of speech parameters whereas channel vocoders involve more conventional analog spectrum analyzer techniques using multiple bandpass filters.

Another curve in Figure 4.2-2, showing derived intelligibility rating for an analog FM system, is included also to show the comparative behavior at lower values of C/kT. For this example, the bandwidth is 25 kHz, a value representative of high quality mobile applications. The conversion of measured FM SNR to intelligibility is more valid for high power levels and less precise at low power levels where the noise introduced by the degradation is nongaussian. Despite this inaccuracy the FM degradation characteristic is comparatively representative of expected performance. Even with the low power impreciseness of the FM curve shown in Figure 4.2-2, digital modulation continues to give acceptable performance well below the relatively sharp cutoff of the FM system.

Several conclusions obtained from Figure 4.2-2 are important to note:

a. The superior performance at low power levels of digital systems over analog systems is demonstrated for several data rates. As power levels decrease the degradation of intelligibility is much more gradual for digital systems. There is no single C/kT value below which performance is considerable degraded as for FM systems. Hence, it is misleading to talk about performance thresholds for digital systems in the same manner as for FM systems.
b. Further, the lower data rate systems are superior to higher data rate systems for certain low power level ranges. This is attributed to the gradual degradation character of CVSD and in some respects

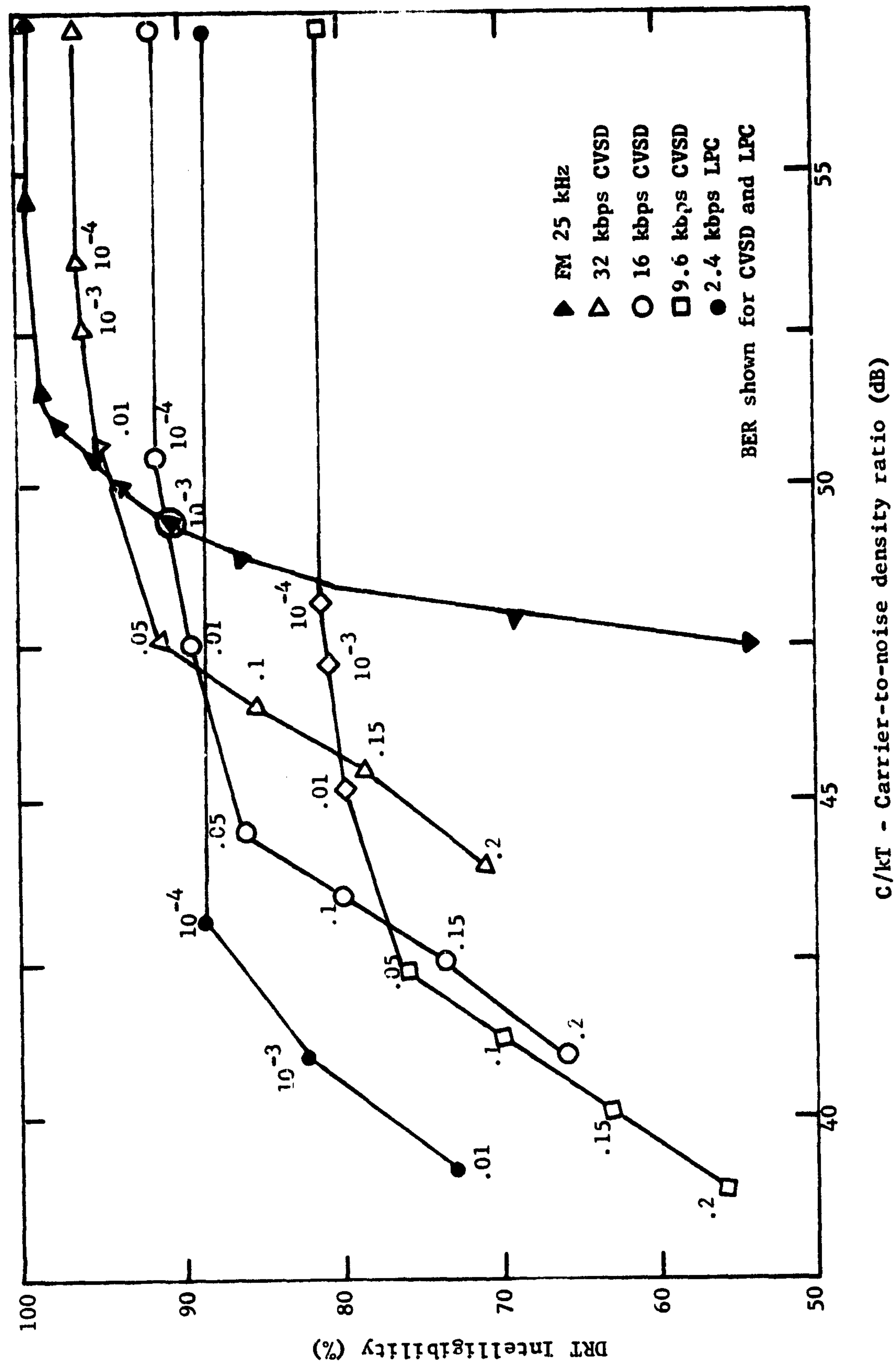

Figure 4.2-2. Intelligibility Performance of Digital and Analog Systems

is similar to the tradeoff between threshold extension and modulation index that exists for FM systems. Although bandwidth is not the prime consideration, lower data rate systems do yield lower BER values for a given power level.

c. It can be seen that there is an optimal data rate for each power level and that data rate generally decreases as power level decreases. Thus, the digital systems are optimal over a very wide power level range because of the flexibility in selecting data rate (and hence bandwidth) using CVSD for voice digitization.

In addition to the intelligibility data, various quality ratings ("how does it sound?" as opposed to "can it be understood?") are shown in Figure 4.2-3 for the CVSD systems. Unfortunately, we are not aware of any similar quality studies for FM systems. The important thing to note in this figure is that speech quality begins falling off at BER values of about 0.01 whereas CVSD intelligibility (from Figure 4.2-2) begins falling off between 0.05-0.1 BER values. Thus, a BER of 0.01 is a better design goal for maintaining acceptable speech quality, as opposed to speech intelligibility, when low power levels are expected.

4.2.3 Signal-to-Noise Ratio Comparisons. A considerable effort was undertaken to compile performance comparisons of the various voice digitization techniques from the open literature. As noted above, there are many signal-to-noise ratio definitions used in the literature and many different ways to measure SNR. The existing information needed to construct a performance matrix for VDT versus SNR is very sparse and considerable effort is necessary to convert the few available results to comparable numbers. In general, the problem is complicated because there are no rules for interpolation and extrapolation, and several inconsistencies in the published results need resolution. Moreover, the availability of good data for intelligibility and quality may indeed make further work on the SNR matrix unnecessary. Therefore, in view of a better alternative, this information is omitted here rather than present a sparse collection of partial results. Suffice it to say that a significant problem exists with the open literature results. In fact, for this particular study objective, message intelligibility in a mobile environment and operation under degraded conditions

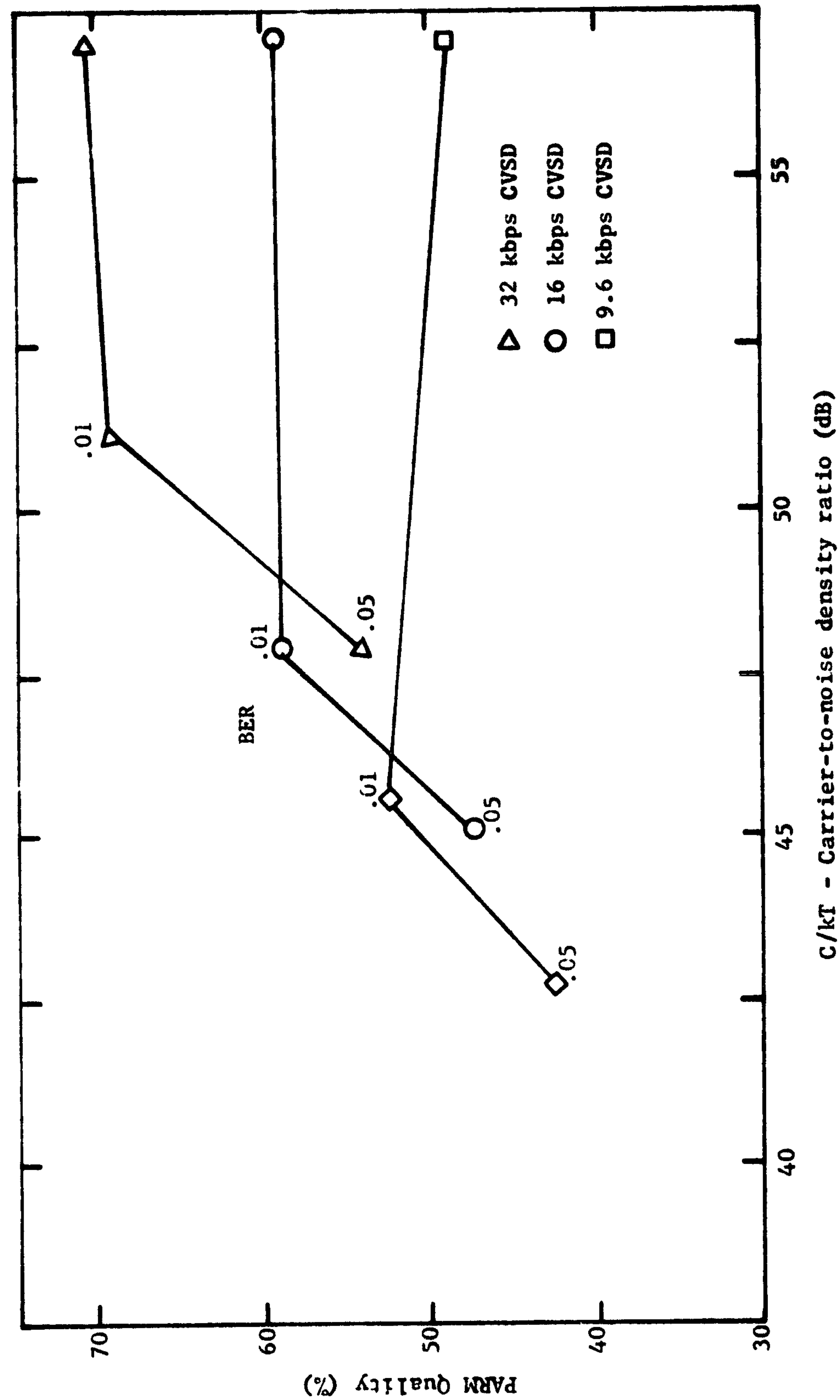

Figure 4.2-3. CVSD Quality Rating

are more important, since SNR is correlated with intelligibility measures for high, but not for low, signal power values. Perhaps the experimental phase should include testing of several voice digitization techniques in operational mobile environments.

4.3 Implementation Considerations

This section summarizes some of the implementation characteristics of the various voice digitization techniques.

Low Rate Vocoders

The typical modern vocoder is implemented in a programmable signal processor of sufficient complexity (usually based on a CPU architecture with about 8,000 words of memory) and speed (between 6 and 10 million multiply and add operations per second). Hence, the cost is in the $10,000 - $50,000 range.

Recently an effort has been started by ARPA to reduce the cost of a vocoder to below $1,000. A combined effort by Texas Instruments Corporation and MIT Lincoln Laboratories is aimed at development of a 19-channel vocoder using charge-coupled device (CCD) - large scale integrated (LSI) technology for the filter banks and single chip microprocessors for the Gold-Rabiner pitch detection. The goals of this development include:

- Intelligibility and quality equal to Belgard channel vocoder
- Power consumption 15 watts
- Volume 1/5 cubic-foot
- Approximately an order of magnitude more complex than CVSD
- Prototypes delivered at the end of 1978 with production a minimum of two years away.

The cost of the vocoders developed under this program still will be expensive. Furthermore, vocoders have severely reduced performance in high background noise environments such as anticipated for mobile stystems. Thus, it does not appear that vocoders are applicable for inclusion in a low-cost

land mobile voice terminal in the near future. Research continues on lower-cost devices (TI has recently announced a synthesizer chip for around $50) and on improving the performance of vocoders in high-noise environments. In the future, vocoders eventually may provide a low-cost, low-bandwidth option.

CVSD Units

CVSD units constructed on LSI Chips exist presently on the commercial market operating at half duplex and cosing approximately $50. Two CVSD chips are needed for full duplex operation (as required here). Some of the suppliers of CVSD chips are:

- o Consumer Microcircuits of America-FX209
- o Motorola-MC 3417 and 3418 (16 pin DIP); external clock, filters, min step-size and gain.
- o Harris-HC-55516 and HC-55532 (24 and 22 pin DIP); one rate, AGC output and idle channel input.

Adaptive Delta Modulators

As noted earlier, there are many ADM algorithms. At least one ADM algorithm has been implemented and has been tested and compared to CVSD. The manufacturer's claim for this implementation states that the performance for operation at 10 kbps is reported to be equivalent to 16 kbps CVSD. However, the knee of the intelligibility versus BER curve occurs at 10^{-2} rather than 10^{-1} as for CVSD. The cost and delivery times estimated by the manufacturer (Deltamodulation, Inc) are as follows:

Quantity	1	200	>1000
Cost	$1800	$350	$50 (Chip)
Availability	Now	3-6 Months	1-2 Yr. ARO

The development of this algorithm has been attributed to Dr. D. L. Schilling of City College, CUNY, New York, N.Y.

4.4 Voice Digitization Recommendations

The above considerations and tradeoffs lead here to the following summary of recommendations for the preferred voice digitization technique to be used in a land mobile voice terminal using satellite communication channels:

a. For lower ranges of C/kT values, digital systems provide more margin than FM Systems of comparable bandwidth.

b. CVSD provides graceful performance degradation as power decreases below the bit error rate curve knee for a given data rate.

c. If data rate is changed to match the available power, a CVSD digital system can operate with acceptable to good performance over the entire range of expected power values for land mobile voice terminal use.

d. CVSD is the most commonly accepted medium-rate technique. Low-cost chips are readily available from several sources. Data rate can be selected on some models within a range of 10 to 32 kbps.

e. DoD is presently investigating alternative techniques for higher quality at 9.6 and 16 kbps.* These techniques are likely to be somewhat more complex and costly than CVSD.

f. Vocoders operating at 2.4 kbps appear prohibitively expensive at present. However, CCD technology eventually may make a channel vocoder feasible. Also, the existence of an LPC speech synthesizer chip (under $50) may indicate future feasibility of an LPC vocoder.

*Reference to established DoD policies and procedures tend to be somewhat elusive, but References 4-13 to 4-17 are indicative of current digital voice technology.

5.0 DIGITAL MODULATION TECHNIQUES

Several candidate digital modulation techniques are examined and compared herein for application to digital mobile communications via satellite. Of those candidates considered, the preferred digital modulation technique appears to be either coherent biphase PSK or quadriphase PSK. This preference is based upon power efficiency of communicating in an austere environment and upon simplicity and low-cost of implementation. This position is reinforced later by link analysis of available link margin and by operational system implementational considerations.

5.1 Candidate Digital Modulation Techniques

For the purposes of this study we have limited the consideration of digital modulation techniques to those types listed below:

- BPSK: Biphase (2) phase shift keying
- QPSK: Quadriphase (4) phase shift keying
- DPSK: Differentially coherent biphase PSK
- MSK: Minimum shift (4) phase keying
- FSK: Noncoherent (2) frequency shift keying
- 8-PSK: Eight (8) phase coherent PSK
- 16-PSK: Sixteen (16) phase coherent PSK

The theoretical performance of digital modulation systems is measured by the bit error rate (BER) and the associated E_b/N_o, signal energy per bit-to-noise power spectral density ratio, required to achieve a prescribed BER. Coherent PSK (i.e. BPSK) with antipodal waveforms, 0^o and 180^o phase shifts, is known to be the most power efficient of all digital modulation techniques, requiring the minimum E_b/N_o for a given BER. Coherent QPSK has the same theoretical performance as BPSK but by its nature requires somewhat more sophistication of implementation in practice and is more sensitive to imperfections. MSK is essentially a special case of QPSK involving controlled phase shifts and resulting in a spectrum more concentrated near the carrier but with its first nulls 50% further out. MSK also requires special demodulation techniques and is not considered to be a serious contender for the low cost digital mobile application. All of the remaining digital modulation techniques listed above are less efficient than BPSK requiring more E_b/N_o for the same BER.

5.2 Performance Comparison

The candidate digital modulation techniques are compared in Table 5.2-1 in terms of required E_b/N_o.

Table 5.2-1. Digital Modulation Techniques-Theoretical Performance Comparison

	E_b/N_o - (dB)	ADDITIONAL E_b/N_o OVER PSK - (dB)			
BER	BPSK/ QPSK	DPSK	FSK	8-PSK	16-PSK
10^{-1}	-0.9	2.9	5.9	4.2	8.8
10^{-2}	4.3	1.6	4.6	4.2	8.8
10^{-3}	6.8	1.1	4.1	4.1	8.7
10^{-4}	8.4	0.9	3.9	4.1	8.7
10^{-5}	9.6	0.7	3.7	4.1	8.7
10^{-6}	10.5	0.6	3.6	4.1	8.7
10^{-7}	11.3	0.5	3.5	4.1	8.7

Since BPSK and QPSK are the most efficient digital modulation techniques, they are used as the reference standard and the required E_b/N_o for specific BER values is shown. The additional (i.e. differential) E_b/N_o over PSK required for these BER values is shown for all the other modulation types. For DPSK and FSK the entries in Table 5.2-1 are exact, for 8-PSK and 16-PSK the entries are approximate. M-ary PSK systems involve specification of an algorithm for decoding the M-ary symbols into a serial binary bit stream. The algorithm assumed here is one in which an error in a M-ary symbol results in equally-likely choosing any one of the remaining M-1 symbols as correct. This assumption leads to a slightly pessimistic requirement. A more realistic M ary detection/decoding scheme would require a few tenths of a dB less E_b/N_o.

A review of Table 5.2-1 points out the heavy penalties paid for the bandwidth conservation advantages of 8 and 16-ary PSK as well as the power inefficiency of FSK. On a power basis the only close competitor to BPSK/QPSK is DPSK which requires exactly 3 dB less E_b/N_o than FSK. Experience has indicated that 8 and 16-ary PSK also are very sensitive to channel imperfections such as band-limiting and nonlinearities with resultant large degradations.

Another means of expressing digital communications performance requirements is the C/kT or C/N_o ratio which is the ratio of signal carrier power-to-noise power spectral density. This is related to E_b/N_o by the formula (in dB)

$$C/kT = E_b/N_o + R_b + L \qquad \text{(dB)}$$

where R_b is the data (bit) rate and L represents any implementation losses associated with the practical realization of the modulation technique.

Figure 5.2-1 shows the theoretical required C/kT for transmission of the various digital modulation techniques as a function of the data rate from 8 to 48 kbps. The curves are plotted for BER from 10^{-2} to 10^{-5}. For DPSK use the curves for FSK and subtract 3 dB, similarly for 8-PSK use the curves for 16-PSK and subtract 4.6 dB. These C/kT curves include a modest implementation loss of 1 dB for BPSK, QPSK, DPSK, and FSK and 1.5 dB loss for 8-PSK and 16-PSK. Implementation losses of this magnitude are typical of high quality equipment but tend to be somewhat higher for low-cost versions. Particularly, we feel that 1.5 dB loss is optimistic for 8 and 16-PSK.

An additional curve has been included in Figure 5.2-1 for a BPSK/QPSK BER of 10^{-1}. This curve illustrates the large additional margin (5.2 dB) available when operating at the end point of the usable range for a CVSD voice-encoded system.

5.3 Error Correction Coding

Forward error correction (FEC) coding can be employed as an adjunct to BPSK and QPSK to improve transmission performance. Typically FEC takes the form of convolutional encoding at the transmitter, often half rate which doubles the transmission rate, and decoding at the receiver using the Viterbi algorithm. The latter may use either hard or soft decisions with differing performance. The performance improvement due to FEC is often referred to as the coding gain and expressed in dB referenced to a specific BER. Coding gain is the reduction in the required E_b/N_o value necessary to achieve a particular BER under the value without coding. It thus represents a power savings at the expense of increased channel transmission rate and, consequentually, bandwidth.

Table 5.3-1 illustrates the coding gains to be expected using half rate convolutional encoding and Viterbi algorithm decoding with both soft and hard decisions. Clearly soft decision decoding is superior to hard decision decoding but entails greater equipment complexity. The principal application of FEC falls in the range of low BER (10^{-5} or lower) and its performace tends to crash rather abruptly at BER higher than 10^{-2}. Also the available coding gain in the vicinity of 10^{-2} and 10^{-3} is diminished tending to make it less attractive for application to the digital mobile communication system.

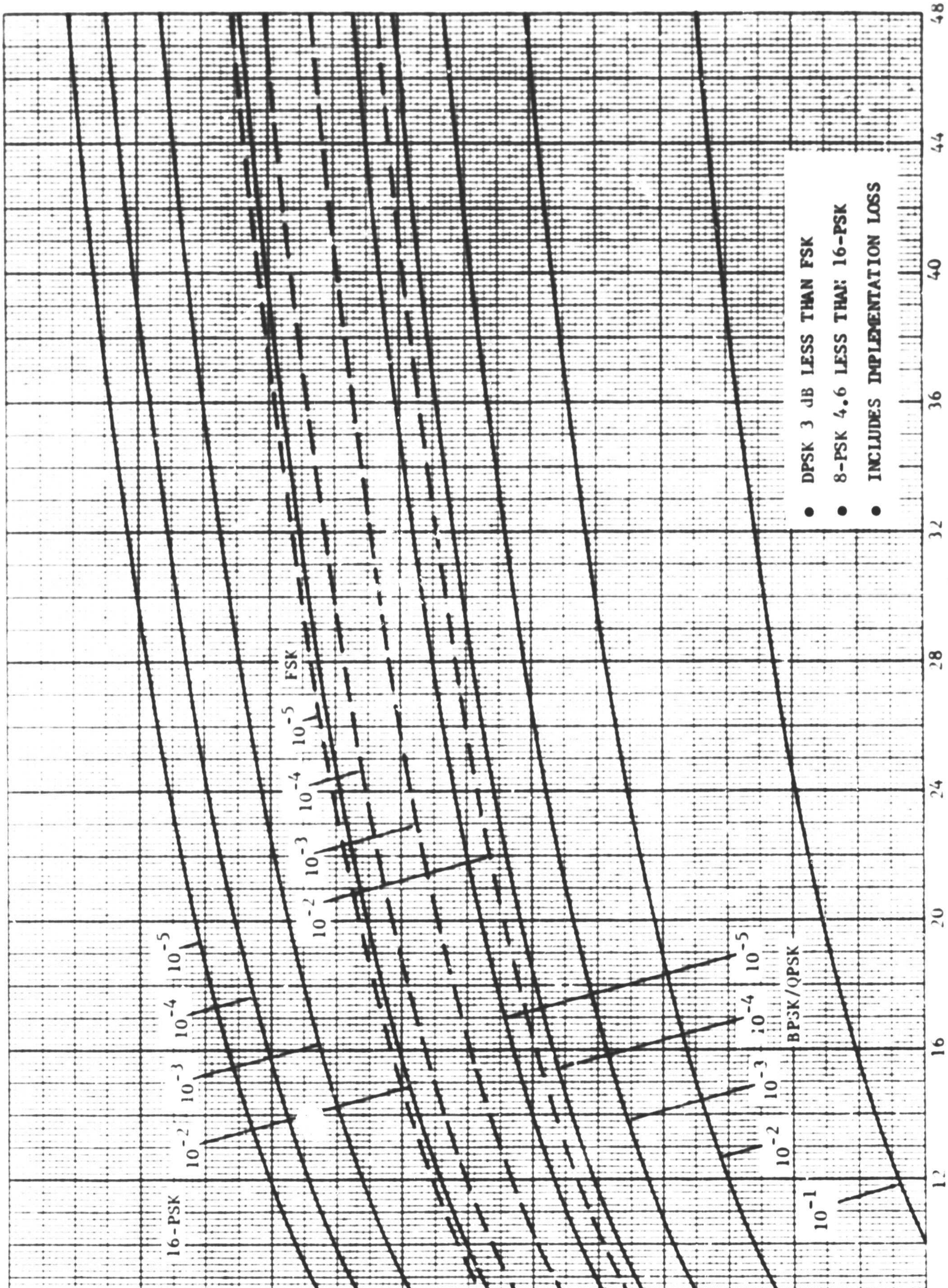

Figure 5.2-1. Digital Transmission Required C/kT

Table 5.3-1. Error Correction Coding Gain for Viterbi Decoding Algorithm

BIT ERROR RATE	SOFT DECISION* (3-BIT A/D)	HARD DECISION
10^{-2}	2.3 dB	0.7 dB
10^{-3}	3.8	1.1
10^{-4}	4.6	2.7
10^{-5}	5.1	3.1
10^{-6}	5.3	3.2

*HALF RATE CONVOLUTIONAL ENCODING

Other considerations associated with the use of FEC are the high cost and various operational aspects such as decoder delay and decoder start-up. Decoder delay is the processing time from the first input bit to the decoder to the time it emerges from the decoder. Decoder start-up is the time necessary to reach steady-state error rate performance. When VOX (voice carrier activation) is used, further complications arise associated with starting and stopping the decoder and its clock in synchronism with the random on-off actions of the VOX operation.

6.0 LINK ANALYSIS

In this section an attempt is made to present a general link analysis of the UHF digital mobile communications system together with parameter variations and evaluations. This approach permits a broad review of systems and techniques before narrowing the field down to a few candidates. In support of this investigation a number of ancillary topics need to be addressed. These topics include intermodulaation crossproducts, voice-spurt-activated carriers, user terminal antennas, and UHF system parameter selection.

The generation of intermodulation crossproducts in the transponder TWTA nonlinearity is discussed along with its detrimental effects. To alleviate this problem a recommendation is made to employ a TWTA backoff of 3 dB as a compromise to achieving reasonable carrier-to-intermodulation ratio and not be excessively wasteful of spacecraft power.

A very important and useful adjunct to SCPC systems is the use of voice-spurt-activated carriers. Appreciable power savings are available by taking advantage of the random nature of speech to key off and on the voice carriers. With a voice activity factor of 35% a 4.5 dB savings in spacecraft power utilization can be realized.

In the following a key point is made that the mobile user terminal antenna is a critical item in the implementation of a mobile network. The requirement to provide omnidirectivity becomes a driving factor in the selection of techniques because it implies the use of rather low gain antennas.

A range of UHF mobile system parameters is adopted here which is expected to be reasonably achievable within the current or near-term technology. Also adopted are two strawman design model mobile systems for more specific analysis and evaluation.

With the use of an omnidirectional antenna it becomes readily apparent that the mobile system is severely down-link power limited. This limitation together with a low-cost goal leads to selection of a preferred techniques combination consisting of BPSK modulation with CVSD voice digitization operating at 16 kbps. Acceptable link margins are provided by this compatible combination because of the inherent graceful degradation characteristic. QPSK modulation could be used also for bandwidth conservation but with increased cost and larger losses.

6.1 Intermodulation Crossproducts

One of the factors limiting link performance of a multicarrier system such as SCPC is the generation of intermodulation (IM) cross products in the satellite transponder nonlinearity. When the SCPC carriers are uniformly-spaced and of equal-strength, the generated IM spectrum is essentially continuous and almost uniformly spread across the transponder bandwidth with a slight peaking at the band center. This IM phenomenon creates a virtual thermal noise floor that effectively limits link quality.

The way to alleviate this source of mutual interference is to operate the transponder TWTA in a quasi-linear mode. Backoff of the TWTA reduces the generated IM level at the expense of throwing away some of the available signal power. This situation has been studied extensively in the past and some results are summarized here. Table 6.1-1 shows the TWTA backoff level and the corresponding carrier power per channel-to-IM cross product density ratio. Results are given for both theoretical calculations and some laboratory measurements. The measured values tend to be somewhat less than the theoretical values.

Table 6.1-1. Transponder Nonlinearity Intermodulation Cross Products

TWTA BACKOFF (dB)	CARRIER/IM RATIO (dB)	
	THEORY*	MEASURED*
0	9.0	
3	15.8	14.9
5	20.9	16.8
6	23.0	17.5

*ASSUMES UNIFORMLY-SPACED, EQUAL-STRENGTH CARRIERS

Our recommendation for this study is to assume a TWTA backoff of 3 dB yielding a carrier/IM ratio in the range of 15 to 16 dB. This compromise allows reasonable link quality while not being excessively wasteful of satellite power.

6.2 Voice-Spurt-Activated Carriers

In a multicarrier system it is reasonable to expect that only a small fraction of the total number of channels will be active simultaneously. This being the case, useful advantage can be made of it through the use of voice - spurt activated carriers (VOX). With this scheme each voice carrier is independently turned off after each pause in the speech waveform. Typically the circuitry is designed to recognize speech pauses longer than 100 to 200 msec and to automatically cut off the voice carrier until the speech waveform returns. VOX operation can provide significant improvement in spacecraft power utilization.

Table 6.2-1 illustrates the spacecraft power savings available with various percentages of voice activity. For this study we have

Table 6.2-1. Voice-Spurt-Activated Carriers

% VOICE ACTIVITY	dB SAVINGS
32	5.0
35	4.5
40	4.0

selected a 35% voice activity as being representative of typical usage and thereby can achieve a 4.5 dB spacecraft power utilization improvement. That is, with VOX a given number of channels can be served with 4.5 dB less power than if all the channels transmitted a carrier 100% of the time rather than only 35% of the time.

It should be noted that VOX operation applies mainly to full duplex systems as envisioned here. Furthermore, VOX operation creates certain receiver reacquisition problems since the carrier is being turned off and on randomly. These problems need to be addressed in an operational system although sensing of the voice pauses is particularly easy in a digital system.

6.3 User Terminal Antennas

The antennas to be employed by the user terminals are of sufficient importance to warrant a separate discussion. In fact, when the development of a digital mobile communication system is undertaken, the user terminal antennas should be regarded as a critical item to be addressed in depth.

For this study we have taken the position that the user terminal antennas should be relatively simple and easily portable. Consequently, two antenna types are considered for possible employment. These candidate antennas are the end-fire helix and the loop over ground plane.

Antennas are easily characterized by their radiation patterns. For example, the helix and loop antennas have representative radiation patterns as sketched in Figure 6.3-1.

The axial mode or end-fire helix antenna has a radiation pattern with axial symmetry and peak gain off the end of the helix. Mounted vertically over a ground plane the peak gain is at the zenith with zero gain at the horizon (ground plane) and uniform symmetry about the axis. As pictured, in any plane through the axis the radiation pattern is a single lobe. The physical dimensions length, diameter, and number of turns determine the maximum gain and beamwidth.

Consisting of a single loop mounted horizontally over a ground plane, the loop antenna has a doughnut-shaped radiation pattern with nulls at the zenith and along the ground plane. In any plane through the axis (i.e., perpendicular to the ground plane) the radiation pattern consists of two lobes with the peak gain at some elevation angle above the horizon. The diameter of the loop in conjunction with the height over the ground plane determines the maximum gain, its associated elevation angle, and the asymmetrical character of each lobe.

Substantial gain is available from the helical antenna, especially for the narrower beamwidths, as depicted in Figure 6.3-2. For the loop antenna lower gain is available and a tradeoff condition exists between maximum gain, its elevation angle, and the elevation angles corresponding to the 3 dB beamwidth points. Selected data is presented in Figure 6.3-3 for the loop antenna showing the maximum gain as a function of loop diameter and of height over the ground plane. Also shown are the elevation angle of the peak gain and the elevation angles of the associated 3 dB beamwidth points.

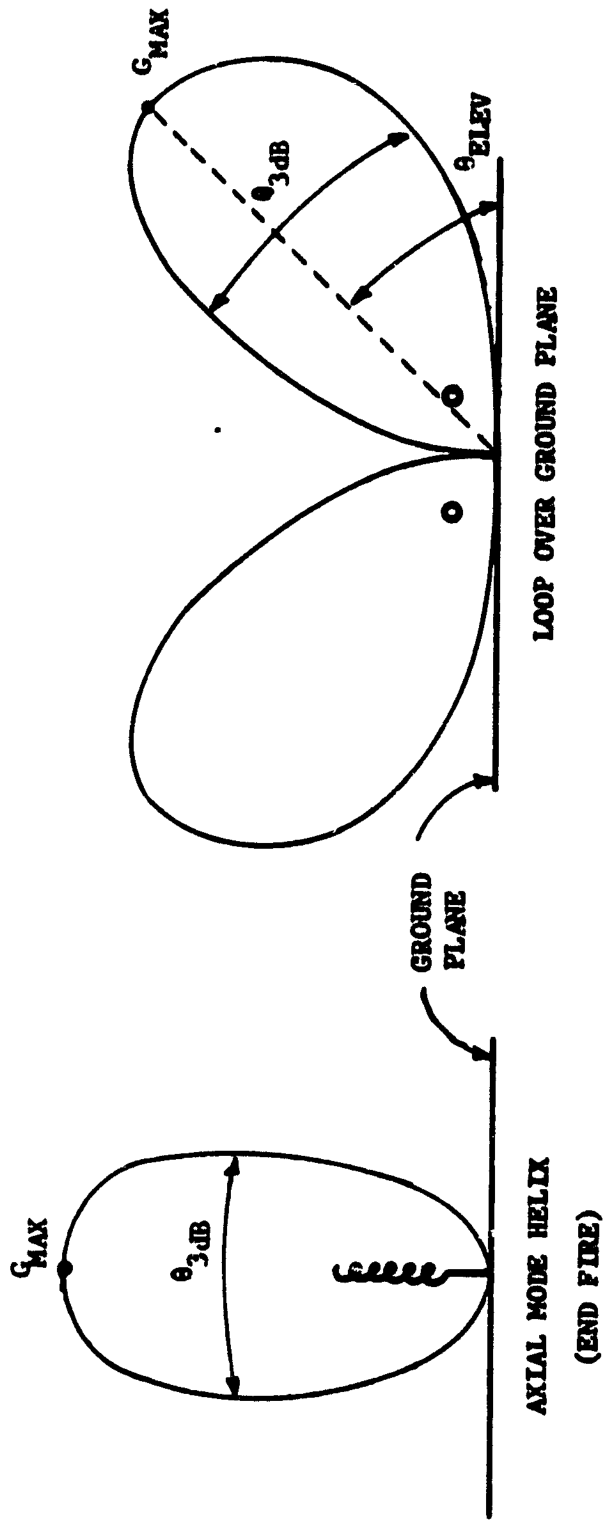

Figure 6.3-1. User Terminal Antenna Configurations

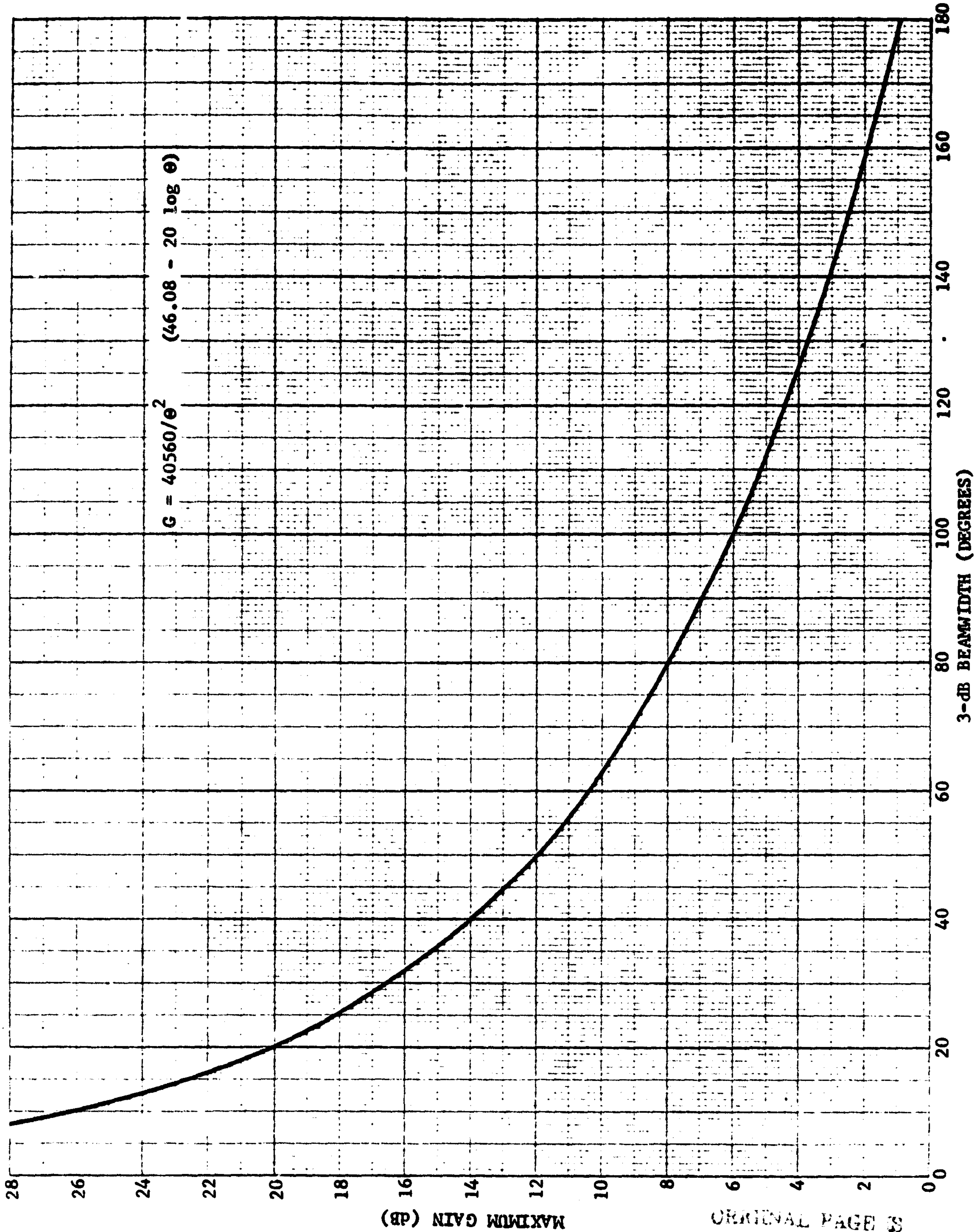

Figure 6.3-2. Axial Mode Helical Antenna Gain/Beamwidth

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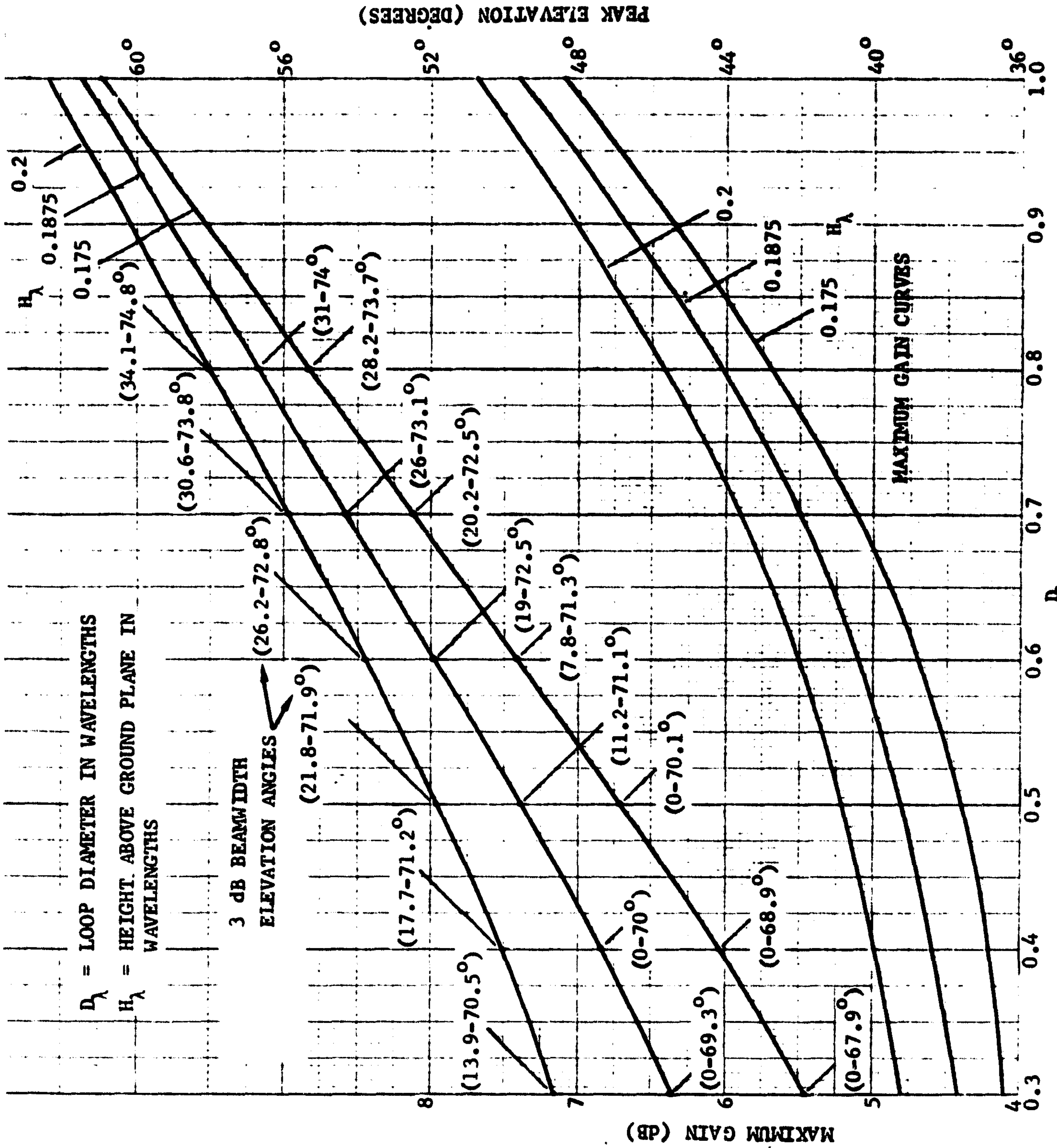

Figure 6.3-3. Loop Antenna Gain/Angular Orientation

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For each of these antenna types we have selected a specific example for further evaluation. Designed for use in the UHF mobile band, approximately 850 MHz, these two antennas are compared in Table 6.3-1 in terms of maximum gain, 3 dB field-of-view (FOV) gain, FOV beamwidth, elevation angles of peak gain and of FOV coverage, and physical dimensions. These example antennas are designated model H (for helix or high) and model L (for loop or low).

Model H antenna offers large gain but with the peak aimed at the zenith and with narrow coverage. Model L antenna offers low gain but the peak gain elevation angle and FOV coverage match well with anticipated CONUS elevation angles of 30° to 50°. The 13.9° to 70.5° FOV elevation angle coverage allows for considerable tilting of the loop antenna platform as might occur in a vehicular installation. The impact of this is that the loop antenna is essentially omnidirectional and well suited to a mobile application. The drawback is, of course, the associated low gain. In contrast, to take advantage of the higher gain helix antenna requires the ability to point the antenna, albeit only coarsely in elevation, continuous azimuthal pointing is necessary for a mobile installation.

Table 6.3-1. UHF Mobile Antenna Comparison

	MODEL H (HELIX)	MODEL L (LOOP)
• MAXIMUM GAIN	13 dB	4.8 dB
• FOV GAIN	10 dB	1.8 dB
• FOV BEAMWIDTH	45°	56.6°
• MAXIMUM GAIN ELEVATION	90°	48.6°
• COVERAGE	90° ± 22.5°	13.9° - 70.5°
• DIAMETER	5.1"/4.0"	4.2"
• HEIGHT	14"/23.7"	2.8"

6.4 UHF Mobile System Parameters

Preliminary to carrying out a link analysis it is necessary to establish a set of system parameters applicable to the UHF mobile system. In achieving this end, a primary objective is to adopt a range of parameter values which are expected to be reasonable within current or near-term technology. The material in Ref. 6-1 relative to public safety, medical, and emergency services at UHF was particularly influencial in this endeavor.

Thus, the objective here is to select a reasonable range of power, antenna gain, and system noise temperature for both the user terminals and the spacecraft. These selected parameters then lead to a range of effective isotropically radiated power (EIRP) and receiver figure of merit expressed in terms of the ratio of receive antenna gain to system noise temperature (G/T) for both the up-link and the down-link. For simplicity of treatment the data presented here is on the basis of only one spacecraft spot beam.

A summary of UHF mobile system parameters is presented in Table 6.4-1 where all antenna gains are stated in terms of FOV (i.e., at the 3 dB beamwidth edge). The data in the upper portion of the table shows the assumed ranges for the earth (user) terminal and for the spacecraft power, antenna gain, and system temperature. These parameter ranges translate into the earth terminal (ET) and spacecraft (S/C) ranges of EIRP and G/T shown in the lower portion of the table. These are in turn used to characterize both the ET and S/C as being of either low, middle, or high quality. For purposes of link evaluation five different combinations are examined later. The five combinations are a low quality ET working with a low quality S/C, a mid ET with a mid S/C, a high ET with a high S/C, and the two crossover cases of low-high and high-low.

In addition to the selected system parameter ranges, two design models have been identified. These are designated as model H (for helix or high) and model L (for loop or low). The only difference between these two design models is the assumed user terminal antenna gain, one uses a high-gain, end-fire helix antenna and the other uses a low-gain loop antenna. Otherwise the two design models have identical parameters. The assumed design model parameters and the resulting EIRP and G/T values are presented also in Table 6.4-1. Again the parameters are stated in terms of the per single S/C spot beam basis.

Table 6.4-1. UHF Mobile System Parameters*

EARTH TERMINAL (ET)			STRAWMAN DESIGN MODELS
POWER	10-32 W	(10-15 dBW)	25 W (14 dBW)
GAIN	0-12 dB	FOV	H: 10 dB L: 1.8 dB FOV
T_{SYS}	500°-800°K	(27-29 dB)	600°K (27.8 dB)
SPACECRAFT (S/C)			
POWER	100-300 W	(20-25 dBW)	200 W (23 dBW)
GAIN	24.5 28.5 dB	FOV	RCV 26.5 dB/XMIT 27 dB FOV
T_{SYS}	700°-1120°K	(28.5-30.5 dB)	900°K (29.5 dB)
FREQ.	823/868 MHz		823/868 MHz

	LOW	MID	HIGH	MODEL H	MODEL L
ET EIRP (dBW)	10	18.5	27	24	15.8
ET G/T (dB/$^{\circ}$K)	-29	-22	-15	-17.8	-26
S/C EIRP (dBW)	44.5	49	53.5	50	50
S/C G/T (dB/$^{\circ}$K)	- 6	- 3	0	- 3	- 3

*DATA PRESENTED ON BASIS OF ONE S/C BEAM

6.5 Link Evaluation

In evaluating link performance all of the previously discussed elements with regard to IM, VOX, antennas, and system parameters are incorporated into this analysis. The discussion deals with only one S/C beam but allows the number of SCPC channels per beam to vary. For simplicity the number of channels per spot beam is either 55, 110, or 220.

The various UHF mobile links examined here are the five combinations and two design models described in Section 6.4. These links are evaluated within the context of a SCPC system with spacecraft power divided equally among the competing channels. A TWTA backoff of 3 dB offset by 4.5 dB of VOX improvement in S/C power utilization is used as mentioned previously. Operating frequencies of 823 MHz on the up-link and 868 MHz on the down-link are employed.

Results of the link calculations are presented in Figures 6.5-1, 2, and 3 for data bit rates ranging from 8 to 48 kbps. The pertinent quantity is the available C/kT per SCPC channel evaluated for each link example with the number of channels variable. These results of available link C/kT are presented by means of overlays upon the required C/kT values for the candidate digital modulation techniques (taken from Figure 5.2-1). This approach permits evaluation of the links against the various modulation techniques. The notation employed in these figures refers to the link quality via a double lettering scheme, each letter standing for low, middle, or high quality (see Table 6.4-1). The first letter of the pair refers to the ET and the second letter refers to the S/C. Strawman design models H and L also are included in this evaluation.

In reviewing Figures 6.5-1, 2, and 3 it is apparent that the assumed UHF mobile system parameter ranges represent a fairly broad range of link conditions. For example, the high quality ET and S/C combination (HH) can support almost any modulation type, data rate, and number of channels per beam. On the other hand, the low quality ET and S/C combination (LL) is not acceptable in any case. These results also point up the value of a high quality ET since the combination HL rates second in all cases. The

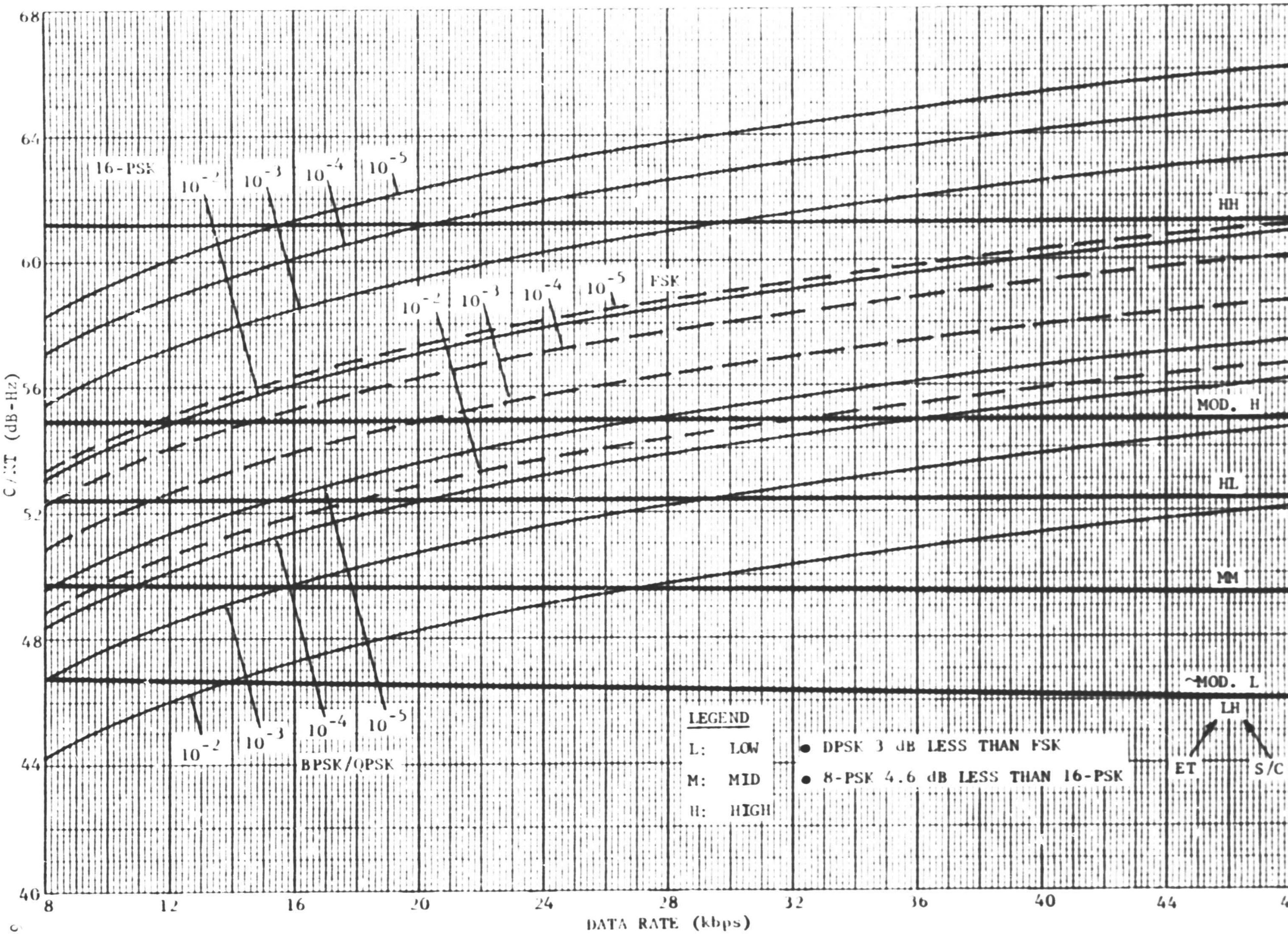


Figure 6.5-1 Available C/kT for System Parameter Variation: 220 Channels

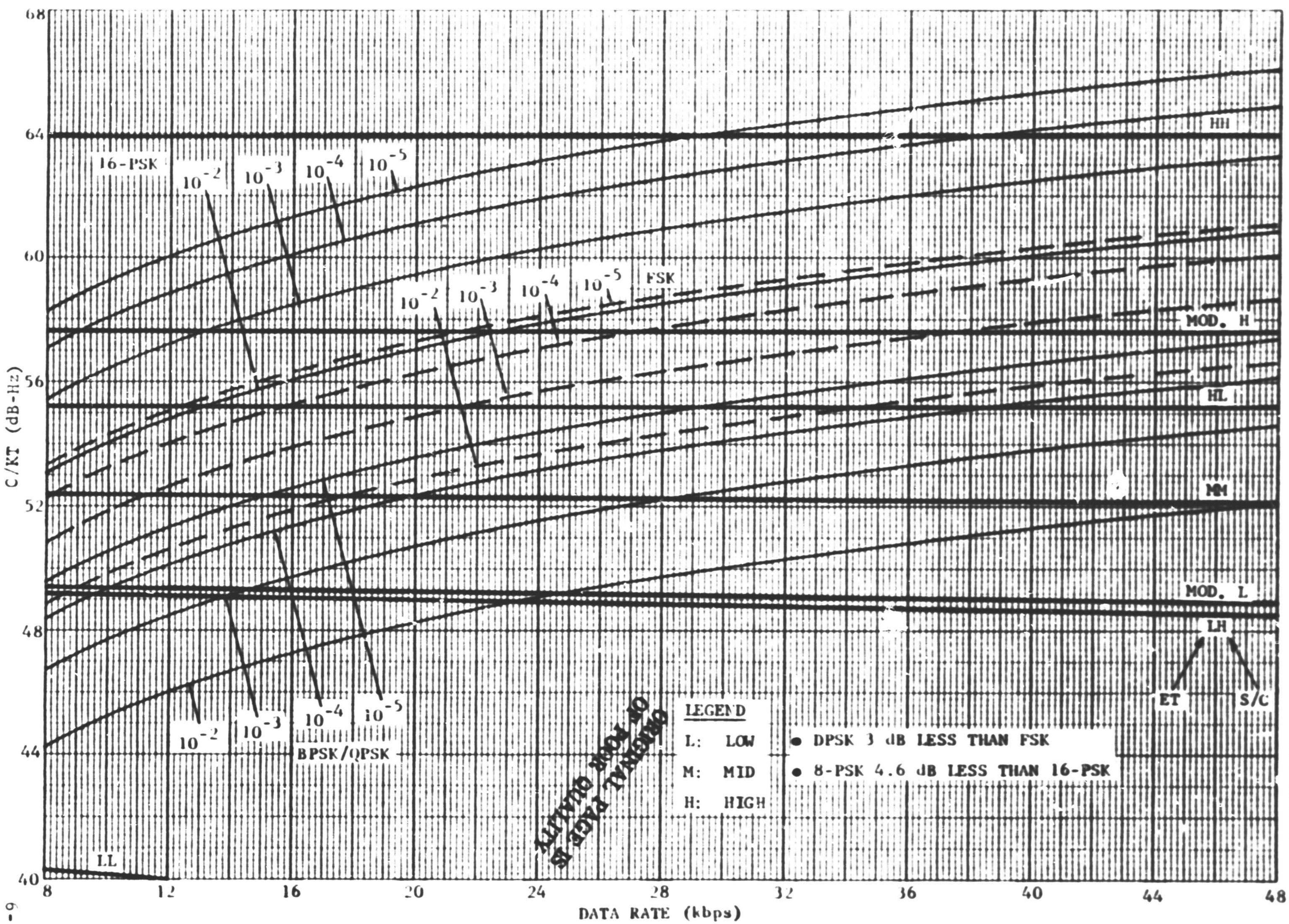


Figure 6.5-2. Available C/kT for System Parameter Variation: 110 Channels

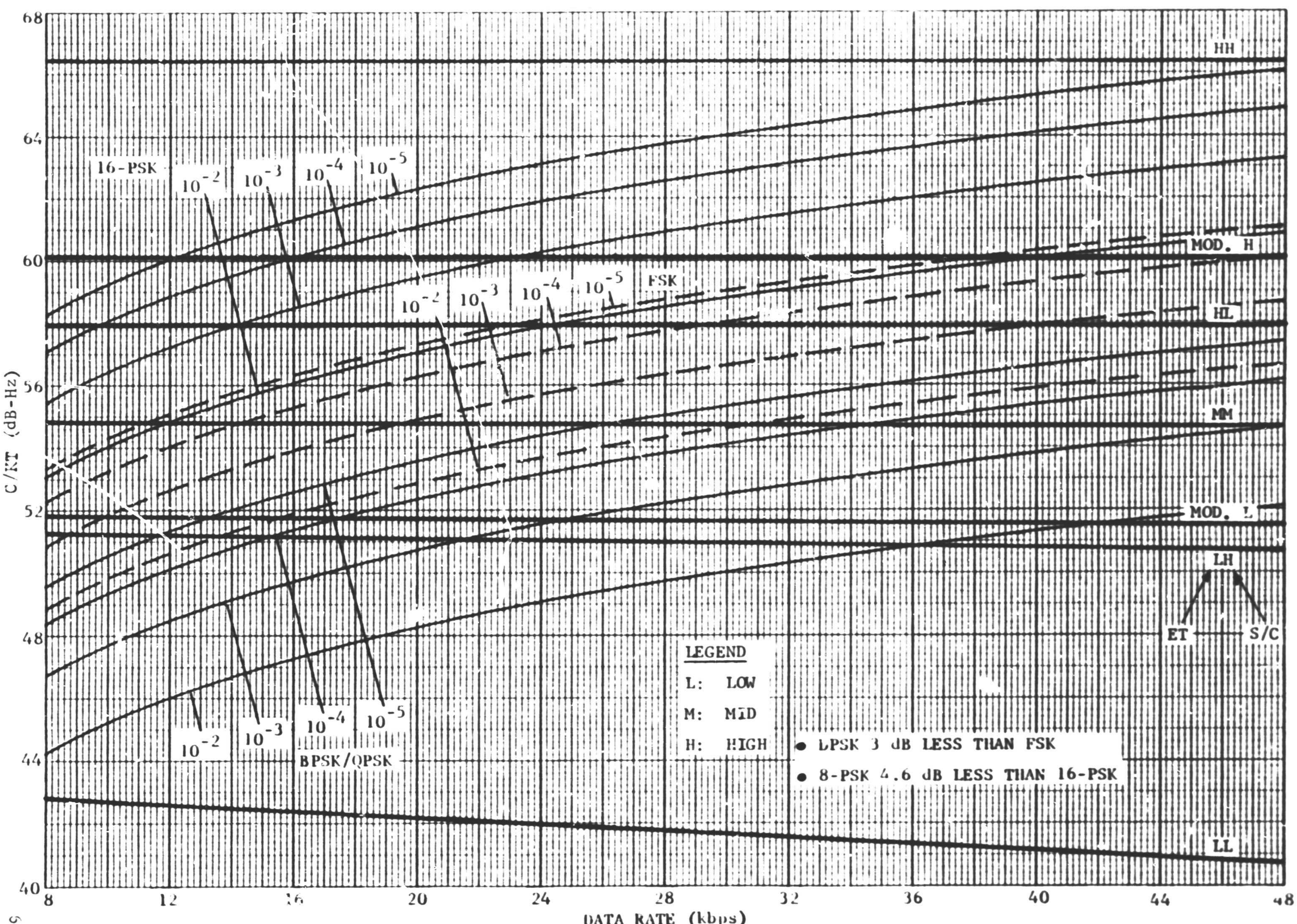


Figure 6.5-3. Available C/kT for System Parameter Variation: 55 Channels

reason for the latter situation is that the link is basically down-link limited and additional antenna gain in the ET improves both the up-link and the down-link.

Evaluation of design models H and L from these figures shows that model H rates better than the link combination HL, but poorer than HH, and model L rates only slightly better than the link combination LH.

The data presented in these three figures represents innumerable trade-offs and comparisons; however, the degree of complexity can be reduced somewhat by focusing on the link margin available for a single modulation technique. For example, take as a reference BPSK/QPSK modulation and a BER of 10^{-3}. The dB difference between the available link C/kT and the required C/kT for the 10^{-3} PSK curve is the available link margin. Figure 6.5-4 coalesces much of the information contained in the previous three figures. Curves are presented in Figure 6.5-4 for the five link combinations and parametrically with 55, 110, and 220 channels per beam. Many of the trade-offs and relative performance evaluations are more easily seen in this new presentation. Extension to other modulation types and BER is possible by cross reference with the previous figures or with Figure 5.2-1.

A similar presentation is made in Figure 6.5-5 for the two design models H and L. Again the reference modulation technique is BPSK/QPSK and the link margin is determined for a 10^{-3} BER. Of the two models, clearly model H offers superior performance due to the large gain of the helix antenna. Measured against the margin for 10^{-3} BER, model L provides questionable performance. However, this observation changes somewhat when the link margin is determined for a 10^{-2} BER reference which requires 2.5 dB less E_b/N_o than 10^{-3} BER. The ordinate scale on the right hand side of Figure 6.5-5 illustrates the available link margin for the new reference. Model L now appears to be somewhat more acceptable than before for at least some range of system parameters.

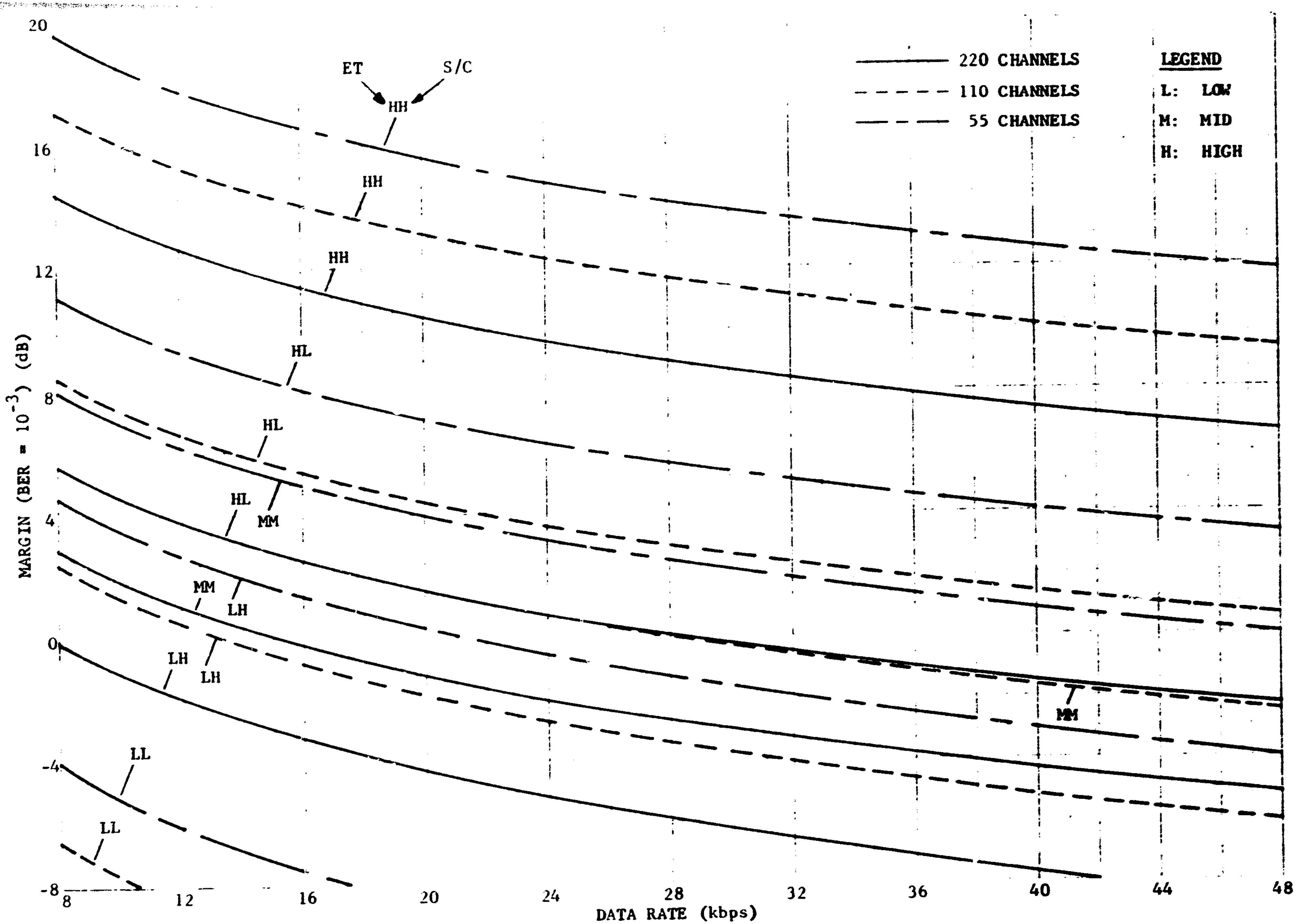


Figure 6.5-4. PSK Link Margin for System Parameter Variation

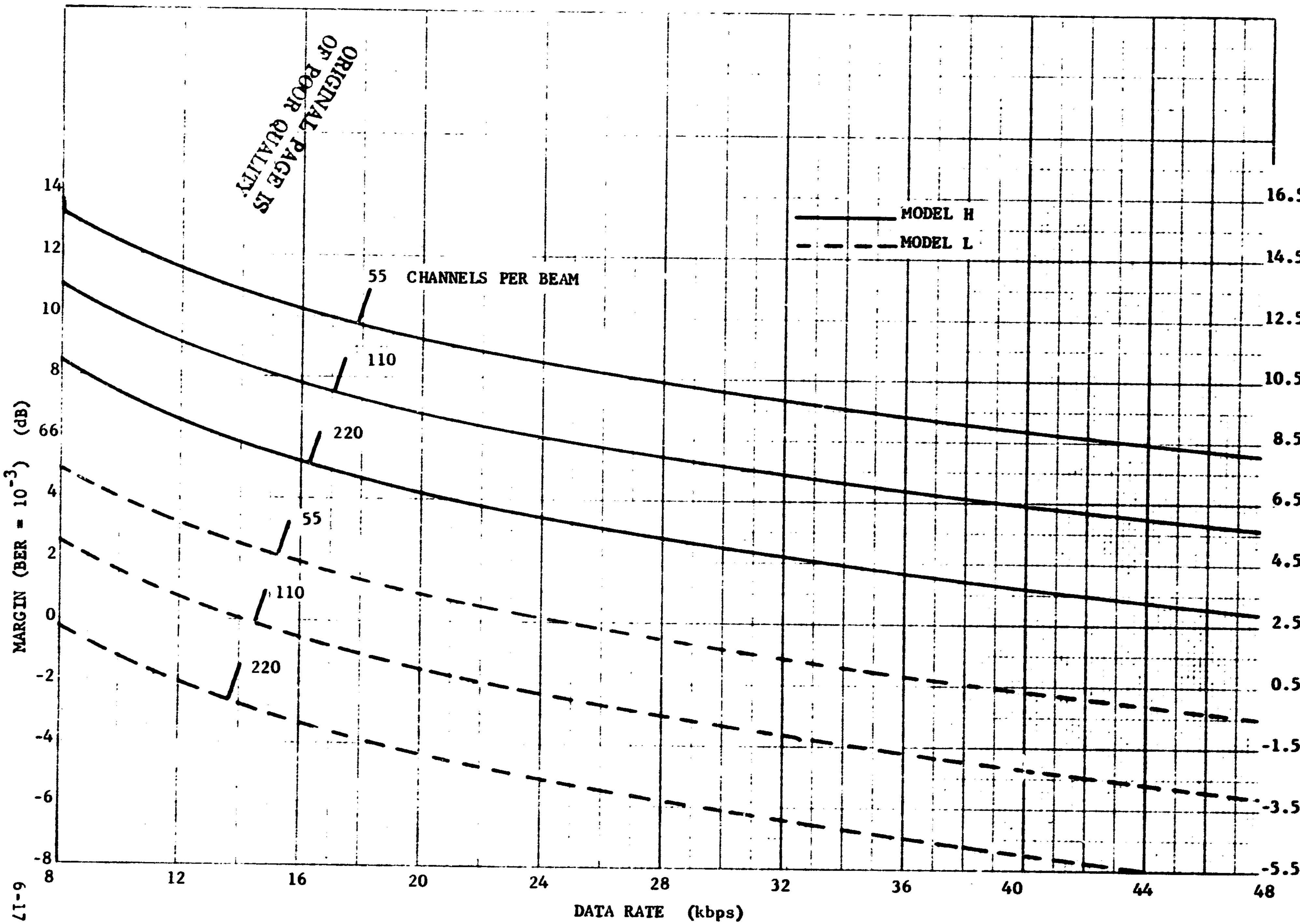


Figure 6.5-5. PSK Link Margins for Design Models H and L

6.6 Techniques Comparison

The preceding evaluation of digital modulation techniques and the associated link analysis has been somewhat abstract and general in nature. However, it has served the purpose of providing the opportunity to examine various parameter tradeoffs and evaluations of techniques. At this point a more fruitful procedure is to narrow the candidate techniques and parameters down to a few specific cases.

A central driving factor is the requirement to provide a low-cost, low burden mobile communications capability which implies an omnidirectional antenna characteristic for the user terminal. This requirement essentially rules out the use of high-gain antennas such as the end-fire helix due to the necessity of pointing to realize its benefits. Therefore, the study must now focus on antenna types such as the loop over a ground plane and adopt the design model L as representative of mobile systems. This position, alone rules out all the digital modulation types except BPSK/QPSK and possibly DPSK, since those others could not be supported with the available C/kT. That is to say, the model L system is severly power limited on the down-link and could only hope to support the barest minimum of 55 channels per beam.

Another point to be made is to discard any use of forward error correction coding. The main reason for this decision is the high cost and attendant complications of FEC. Moreover, the use of FEC in this application is not very effective because of the high error rates involved in the mobile voice communication system.

As concluded in Section 4.0 the preferred voice digitization technique is CVSD because of low cost, good tolerance to errors, and especially its very graceful degradation. In fact, the latter characteristic is the key to achieving a viable digital mobile communication system. A competitor is linear predictive coding (LPC) with somewhat less tolerance to errors and less graceful degradation. The main advantage of LPC is its ability to operate at very low data rates, but at the present time the cost is prohibitive.

The above ensemble of factors and considerations have been incorporated to enable comparison of four digital mobile system techniques. Table 6.6-1 presents a comparison of these four finalists. The digital modulation techniques are BPSK/QPSK and DPSK, the voice digitization techniques are CVSD at

Table 6.6-1. Digital Mobile System Techniques Comparison

DIGITAL MODULATION TECHNIQUE	BPSK/ QPSK	DPSK	BPSK/ QPSK	DPSK
VOICE DIGITIZATION TECHNIQUE	CVSD	CVSD	LPC	LPC
INFORMATION DATA RATE (kbps)	16	16	2.4	2.4
DESIGN POINT BER	10^{-3}	10^{-3}	10^{-4}	10^{-4}
THRESHOLD BER	10^{-2}	10^{-2}	10^{-3}	10^{-3}
USABLE END POINT BER	10^{-1}	10^{-1}	10^{-2}	10^{-2}
DATA RATE (dB-Hz)	42.0	42.0	33.8	33.8
MODULATION DESIGN POINT E_b/N_o (dB)	6.8	7.9	8.4	9.3
NOMINAL IMPLEMENTATION LOSS (dB)	1.0	1.0	1.0	1.0
REQUIRED C/kT (dB-Hz)	49.8	50.9	43.2	44.1
AVAILABLE LINK C/kT* (dB-Hz)	51.8	51.8	51.9	51.9
DESIGN POINT LINK MARGIN (dB)	2.0	0.9	8.7	7.8
ADD'L. AT THRESHOLD BER (dB)	2.5	2.0	1.6	1.4
THRESHOLD LINK MARGIN (dB)	4.5	2.9	10.3	9.2
ADD'L. AT USABLE END PT. BER (dB)	5.2	2.9	2.5	2.0
USABLE END PT. LINK MARGIN (dB)	9.7	5.8	12.8	11.2

*MODEL L PARAMETERS WITH 55 CHANNELS PER BEAM

16 kbps and LPC at 2.4 kbps. In Table 6.6-1 we have introduced the concept of a design point BER at which the system is expected to operate at or be better than. Also introduced is the threshold BER at which the system can degrade to without serious loss of intelligibility or quality. Another fallback point is the BER corresponding to further degradation at which the system is judged to have reached the end of its usable range. For each candidate system, link margins are calculated corresponding to the above three operating points.

Examination of Table 6.6-1 reveals that the LPC systems offer the greatest link margins for all three operating points. However, the cost of presently available LPC units makes this choice unacceptable for the low-cost mobile application. Most likely, as both the technology and the market for LPC develops this approach will become the favorite of the future.

Of the two CVSD systems, the one employing DPSK yields rather low link margins. Thus, it becomes apparent that for the current and near-term technology, the best choice of system techniques is CVSD with either BPSK or QPSK. This selection offers a modest link margin at the design point BER and considerably more link margin before reaching the threshold BER and still more at the end point BER of the usable range. Use of either BPSK or QPSK modulation provides a total of 7.7 dB additional margin below the 10^{-3} BER design point before reaching the end of the usable intelligibility/quality range for CVSD voice digitization. Including 1 dB for nominal implementation loss, this indicates that BPSK and QPSK systems can operate down to a C/kT of 42.1 dB-Hz with a data rate of 16 kbps. Futhermore, recall that these link margins are calculated for the FOV condition for both the user terminal and for the spacecraft. Therefore in practice these worst case conditions should apply only to a fraction of the user population and larger link margins (better performance) should be expected for the majority of users.

As for a choice between BPSK and QPSK modulation, we must examine factors relating to practical implementations. QPSK, of course, offers the advantage of a 50% savings in required channel bandwidth and in some applications this may be the overriding consideration, although probably not so here. On the other hand, QPSK entails greater cost and complexity than BPSK as well as greater sensitivity to channel impairments and imperfections. Generally speaking, because of this sensitivity larger implementation loss is associated

with QPSK than with BPSK. This is especially true in low cost applications where 1.5 to 2.5 dB implementation loss is typical for BPSK and 2 to 3 dB or more is typical for QPSK. VOX operation also impacts these considerations by further aggravating the sensitivity problem and leading to increased cost, complexity, and implementation loss for QPSK. This issue is related to the more severe (much greater than just a factor of 2) requirements associated with acquisition and reacquisition of the QPSK receiver with voice-spurt-activated carriers (i.e., carrier reconstruction loop problems). After reviewing these factors and in the absence of a strong bandwidth constraint. we recommend adoption of BPSK over QPSK in the interest of low cost, simplicity, minimal implementation loss, and relaxed VOX requirements.

6.7 Low-Rate Data Transmission

In addition to the primary digital voice function, a requirement of the mobile terminals is to provide for low-rate data transmission, most likely in the 300 bps to 2.4 kbps range. The purpose of this secondary function is to provide a mobile communications public service data transfer capability. Examples of the use of this type of function include data transfer for computer data bank files of patient records, laboratory data, and medical practice information; for transferring criminal history files, criminal justice planning, fingerprints, and mug shots; data transfer of government personnel records, data banks, and library services; and transmission of general facsimile, printed, or graphical information. As contrasted to voice communications, with low-rate data transmission for public services of the above type the emphasis is on the accuracy of the data transfer. Therefore, low-rate data applications typically involve much lower bit error rates than voice communications, which can tolerate frequent errors due to the redundancy of speech and the characteristics of the ear.

Generally speaking, if the communications system is designed to adequately handle the higher-rate voice transmission, there is no significant impediment to low-rate data transfer. The simple reduction of the data rate alone ensures improvement in performance on the basis of the available C/kT value versus the lowered required value. This implies higher available link margins for the low-rate data.

For the purpose of this discussion, we take as a baseline the model L system parameters using BPSK modulation to take advantage of the compatability of the information types (i.e., digitized voice and data). Table 6.7-1 shows the available link margins for low-rate data at several different bit error rates and at two typical data rates.

Table 6.7-1. Low-Rate Data Link Margins (dB)*

	300 bps			2.4 kbps		
	BER			BER		
Channels/Beam	10^{-4}	10^{-6}	10^{-7}	10^{-4}	10^{-6}	10^{-7}
55	17.7	15.6	14.8	8.7	6.6	5.8
110	15.3	13.2	12.4	6.2	4.1	3.3
220	12.5	10.4	9.6	3.5	1.4	0.6

*BPSK Using Model L Parameters

In evaluating these results it is apparent that the low-rate data function can be achieved at even very low bit error rates without the need for any aids such as forward error correction. Even the 220 channels per spot beam mode at a BER of 10^{-7} has some margin in a worst case situation.

A complete description of the operational features of implementing the low-rate data capability in conjunction with the digital voice modem is presented later in Section 8.4. Needless to say, one of the bonuses of the use of digital voice modulation is the compatability and ease of providing for digital data transfer.

The low-rate data function differs from the primary voice function in that data transfer transmissions tend to be of long duration and continuous in nature. This type of transmission characteristic indicates that the expected VOX spacecraft power utilization improvement may be somewhat diminished for low-rate data. However, since each spot beam contains a large number of carriers operating primarily with voice, the effects of a few low-rate data carriers should not seriously reduce the overall VOX improvement for the spot beam as a whole. Thus, we expect the performance levels indicated in Table 6.7-1 to be fairly representative for low-rate data transmission.

7.0 DEMAND ASSIGNMENT MULTIPLE ACCESS

In order to provide services to a large number of users with only limited satellite resources, a Demand Assignment Multiple Access (DAMA) system will be needed. A DAMA system is required to select signal paths, satellite channels, between users on an "as needed" basis. In this respect, the DAMA system is analogous to a telephone switch, except that it assigns frequency slots in the satellite, rather than crosspoints in the telephone network.

Since the DAMA system acts as a telephone switch, it should be transparent to the user. That is, when a user places a call, he should perform the normal calling (dialing) functions and receive the normal call progress information to the maximum extent possible. Thus, the DAMA system should provide the normal call establishment features including dial tone, ringback signals, busy signal, and release tone. In addition to meeting the above user-oriented requirements, the DAMA system should have the following desirable features:

a. Minimize post dialing delay
b. Minimize number of signalling channels (overhead)
c. Minimize overall system cost
d. Capable of system clearing when overloaded
e. Capable of priority assignment and break-in

With these features in mind, a centralized DAMA system is considered, employing random access for call requests and a broadcast mode for channel assignments and network control. A centralized DAMA system uses a single earth station for network control, referred to as the Master Control Station (MCS). To minimize the total network cost, each mobile earth terminal is provided with only one modem which is used for both signalling with the MCS and for communicating with the other remote terminals. In keeping with the shared-modem concept for cost/complextity reduction, we assume the DAMA system operates at the same data rate as the voice channels. The transmission rate of 16 kbps is selected to simplify the modem design such that the same modem can be used for digitized voice as well as for DAMA messages without any adjustment.

In Section 8 a complete operational DAMA system is described in depth having the features indicated here. In addition the operational description includes a step-by-step procedure for a typical call sequence illustrating the transparency of the system to the user. For these reasons the discussion here concentrates on the analysis and tradeoffs of such a DAMA scheme.

To carry out an analysis of a DAMA system requires the assumption of a specific traffic model. The model employed here is based on the call statistics of Section 3, but for illustration limits the number of mobile terminals to 10,000. Table 7.0-1 shows the parameters selected for the DAMA traffic model.

Table 7.0-1. DAMA Traffic Parameters

• TERMINAL POPULATION	10,000
• TOTAL AVERAGE CALL ORIGINATIONS	184.9 CALLS/SEC.
• AVERAGE CALL DURATION	11.47 SEC./CALL

7.1 Common Signalling Channel Access Tradeoff

For the centralized DAMA system, it is necessary for the remote user terminals to communicate with the MCS on the Common Signalling Channel (CSC). There are two common categories of techniques: polling and random access. (Although we have preselected the random access approach, this discussion points up the tradeoffs involved.)

In the polling technique, the MCS individually polls each remote terminal sequentially. The remote terminals respond only when polled by MCS command and no mutual interference can occur.

For the random access case, each user terminal transmits only if it has a message to send. Within the random access category, there are two subcategories, the slotted and the unslotted cases. For the slotted case, the time scale is divided into slots either based on the last transmission or established by a master-start timing burst. The user terminal is allowed to transmit only at the begining of each slot. For the unslotted case, the user

transmission is completely at random. Hence, for unslotted random access, it is possible to have two or more terminals transmitting at the same time, thereby interfering with each other. In this case, the messages would not be received correctly by the MCS and no reply would be sent to the remote terminals. However, after a certain random time delay, the user terminals, not having received a reply, retransmit their access request messages.

There are a number of parameters to be considered in comparing the polling and random access techniques. These are: CSC data rate, number of user terminals, traffic loading of the network, and time delay of the call process. Of these several parameters, only time delay would be noticeable by the user. Thus, it is used as the basis for comparison.

For a system that is not overloaded (i.e., no backlog at user terminals for the polling technique and no blocking condition for the random access), the time delay for the polling technique is a function of the number of terminals in the network while the time delay for the random access technique is a function of the total network traffic requirement.

7.1.1 Polling CSC Time Delay. The average time delay, $\bar{d}_{poll}$, between the time that a call origination request is ready for transmission by a user terminal and the time the master station receives the request is:

$$\bar{d}_{poll} = \frac{1}{2} \left[T_r + (N_s - 1) (T_p + T_g) \right]$$

where T_r = satellite two-way round-trip time delay

N_s = number of terminals being polled

T_p = message length

T_g = guard time between messages

The above expression assumes that the backlog of messages at the user terminal is essentially zero, which is the present case where each remote terminal only has one modem.

Using a message length* of 72 bits and a transmission rate of 16 kbps, T_j = 4.5 msec. A reasonable guard time is approximately 1 msec for transmission through a synchronous altitude satellite having a fair knowledge of the location of the user terminal. For mobile terminals covering a wide area, a longer guard time is required. The two-way round trip time delay is 0.54 sec for synchronous altitude satellite communication, thus, the average time delay is:

$$\bar{d}_{poll} = \frac{1}{2} \left[0.54 + (N_s - 1)\ (0.0055) \right] \quad \text{sec.}$$

If it is desired to have an average time delay of, say, 1 second, each polling channel is limited to poll 266 user terminals. To poll 10,000 user terminals, 38 satellite channels are required. Including the 38 broadcast channels, a total of 76 satellite channels are needed for signalling using the polling technique.

7.1.2 Random Access Time Delay. As mentioned previously, for random access, there are two potentially useful versions: slotted or unslotted. For the slotted case, the timing for these slots is usually obtained by monitoring the transmissions of the master station to acquire a reference timing burst. Similar to the polling case, the time slot should include a guard time to account for the uncertainty of the arrival time to the satellite. For the same message length and transmission rate as mentioned above, the slot time should be 5.5 msec. The probability of collision for the slotted case is equal to the probability that two or more messages are transmitted during the same time slot.

For the unslotted case, messages are transmitted as soon as they are ready. The probability of collision is equal to the probability that two or more transmissions are started within two adjacent message time intervals (9 msec).

For the case of an unlimited number of retransmissions until acknowledgement of a successful transmission is received, the average delay $\bar{d}$, including the one-way transmission delay of $T_r/2$ (0.27 seconds), between the transmission and successful reception of the message at the master station is:

*Typical message formats are described in Section 8.5.

$$\bar{d}_u = T_r \left(e^{2G} - \frac{1}{2}\right) \quad \text{(unslotted case)}$$

$$\bar{d}_s = T_r \left(e^{G} - \frac{1}{2}\right) \quad \text{(slotted case)}$$

where G is the fractional utilization of the total channel capacity for both original and retransmitted messages. The value of G may be calculated by solving the transcendental equation which relates G, the actual channel loading, and S, the offered load (equal to throughput for a stable channel):

$$S_u = G\, e^{-2G} \quad \text{(unslotted case)}$$

$$S_s = G\, e^{-G} \quad \text{(slotted case)}$$

For these equations, the maximum throughput, S_u and S_s, of the channel can be calculated at 18.4% and 36.8%, respectively. However, to minimize the delay and to maintain a high degree of stability, the channel should be designed to operate at approximately 10% and 20% respectively. For $S_u = 10\%$, G is calculated to be 0.1296. For $S_s = 20\%$, G is calculated to be 0.2592. The average time delays for the two cases are:

$$\bar{d}_u = 0.43 \text{ sec.}$$

$$\bar{d}_s = 0.43 \text{ sec.}$$

With throughputs of 10% and 20% for the unslotted and slotted cases, the requirements for random access are summarized in Table 7.1-1.

Table 7.1-1. Random Access Requirements

	UNSLOTTED CASE	SLOTTED CASE
MESSAGE OR SLOT TIME	4.5 msec	5.5 msec
NUMBER OF MESSAGES OR SLOTS	222/sec	182/sec
THROUGHPUT	22.2/sec	36.4/sec
TOTAL NUMBER OF MESSAGES TO BE TRANSMITTED *	554.7/sec	554.7/sec
NUMBER OF CSC CHANNELS REQUIRED	25	15

*Requires 3 calls per call origination

Since the messages from the MCS to the user terminals are time division multiplexed with no chance of collision, the broadcast channel can operate with close to 100% throughput. At 80 bits per message and a 16 kbps data rate, the MCS can transmit 200 messages per second. For 184.9 call originations per second for the entire network and 4 messages to the user terminals per call setup/release, a total of 4 broadcast channels will be required. Thus, the total number of signalling channels required are 29 and 19 for the unslotted and slotted cases, respectively. Taking into consideration an uneven message traffic loading distribution among the groups of user stations assigned to the specific CSC frequencies, a few more signalling channels may be needed. However, the total number of signalling channels required would still be less than the polling case.

7.2 DAMA Access Tradeoffs

The two DAMA access schemes, polled versus random access (slotted and unslotted), using the traffic model described here at 16 kbps transmission rate, are compared in Table 7.2-1. Clearly the polled access scheme leads to a requirement for a large number of channels, 2 to 4 times more channels than for random access. This factor alone makes random access more attractive for the mobile terminal application.

Of the two random access schemes, the unslotted approach requires about 50% more total satellite channels (29 versus 19) dedicated to DAMA functions and on this basis appears less desirable than slotted DAMA. However, a slotted approach requires a greater degree of cooperation between the network users, and the MCS, than does the unslotted approach. To implement the slotted scheme requires each user to monitor the broadcast channel and to adjust his access request transmissions to fall within the available access slots. This requirement places an unwanted burden, in terms of cost and complexity, upon the user terminals. Therefore, to achieve simplicity of design and low-cost, we recommend adoption of the unslotted random access scheme for the mobile terminal system.

Table 7.2-1. DAMA Common Signalling Channel Access Tradeoff

		COMMON SIGNALLING CHANNEL		
	ACCESS TIME DELAY	ACCESS CHANNELS	BROADCAST CHANNELS	TOTAL CHANNELS
FIXED ACCESS/POLLED	1 SEC	38	38	76
RANDOM ACCESS/SLOTTED	0.43 SEC	15	4	19
RANDOM ACCESS/UNSLOTTED	0.43 SEC	25	4	29

8.0 OPERATIONAL SYSTEM DESCRIPTION AND COST ESTIMATE

This section reports on the results of incorporating the fundamental study assumptions together with the conclusions of the voice digitization and digital modulation techniques tradeoffs to form an operational digital mobile communications satellite system configuration. The operational system description presented here includes the system concept, hardware considerations, and selection of system design parameters and operational procedures to provide a capability for mobile voice communications, low-speed data operation, and demand assignment multiple access. In addition to the operational description, a cost model of the mobile terminals has been developed exploiting the application of LSI technology wherever possible for quantities of 10 to 10,000 terminal units. The design description of the mobile terminals was developed in sufficient detail to a level that permits realistic cost estimates of the operational system.

Basically the operational system described here employs biphase PSK digital modulation in conjunction with continuously variable slope delta modulation (CVSD) voice digitization operating at a 16 kbps data rate. This combination of modulation and voice encoding techniques offers reasonable quality system performance together with low terminal complexity and a corresponding low cost. Options are provided for handling data services at any rate up to 16 kbps plus the capability of demand assignment multiple access for large terminal populations.

A unique feature of the BPSK data demodulator described here is the proposal to employ a virtually all-digital implementation. This design is a digitally-implemented version of the classical analog Costas PLL and offers the distinct advantage of being relatively inexpensive to implement compared to conventional analog circuitry. Moreover, this all-digital approach has been shown to achieve measured performance to within only 1.2 dB of theory and to be operable at rather low received signal levels.

Another departure from the conventional approach is the proposed satellite configuration and the associated traffic/frequency plan. This operational system concept involves a multibeam antenna satellite with a large number of down-link spot beams operating in the mobile UHF band and a single, common global up-link receive antenna operating a L-band. The use of L-band and UHF dual frequency bands

for transmit and receive is forced by the lack of sufficient frequency space at UHF to accommodate the large number of channels anticipated for full CONUS coverage. In addition, the large number of service regions require the use of down-link spot beams to provide sufficient satellite EIRP.

A comprehensive summary of the operational system parameters and specifications is presented here in Table 8.0-1 and a pictorial representation is illustrated in Figure 8.0-1.

Cost estimates were developed for operational mobile terminal designs in quantities of 10, 100, 1000, and 10,000 units. For the lower quantities the application of DAMA operation is judged to be economically unfeasible. A cost summary of mobile terminal cost estimates without low-speed data or DAMA options appear below:

Number of Units	Cost per Unit
10	\$64,380
100	\$20,946
1,000	\$ 5,756
10,000	\$ 3,947

These cost estimates include both recurring and non-recurring costs for mobile terminals with voice capability only. For large terminal populations, such as 10,000, the additional cost of providing both low-speed data and DAMA capability is expected to be only \$610 per terminal for a net terminal cost of \$4,557.

8.1 Operational System Concept

The operational version of the mobile digital communications network is assumed to work in both the UHF mobile radio band and in an unspecified frequency allocation at L-band. Up-link and down-link frequencies are considered to be separated as much as possible in order to avoid excessively severe receive filter requirements. The up-link frequency band would be at L-band, where more bandwidth is available, and the down-link frequency band would be in the 806 to 890 MHz UHF mobile band. The entire CONUS area is divided up into a number of regional service areas, possible as many as 80. Some portion of two or more adjacent regional areas may lie in a common overlap area, especially along the crowded eastern seaboard. Each regional area in CONUS is served by a separate satellite spot beam in order to provide sufficient satellite EIRP. Each mobile ground terminal can communicate directly with any other mobile ground terminal by means of an appropriate SCPC

Table 8.0-1. Operational System Parameters and Specifications

ITEM	SPECIFICATION	COMMENT
FREQUENCY PLAN	FDMA-SCPC	DUAL BANDS
• UP-LINK	L-BAND (UNDEFINED)	CONTIGUOUS FDMA
• DOWN-LINK	UHF MOBILE	RE-USE FDMA
SATELLITE CONFIGURATION	MULTIBEAM ANTENNA	FREQUENCY TRANSLATES IN MULTIPLES OF 4 BEAMS
• UP-LINK ANTENNA	GLOBAL RECEIVE	CONUS COVERAGE
• DOWN-LINK ANTENNA	SPOT BEAM TRANSMIT	REGIONAL COVERAGE
NUMBER OF CHANNELS (FULL DUPLEX)	56/FREQUENCY BAND, UP TO 80 BANDS	2 CHANNELS/BAND RESERVED FOR DAMA
BANDWIDTH	45 kHz/CHANNEL 3.02 MHz/BAND	SPECTRAL SHAPING (?) 0.5 MHz GUARD BAND
VOICE ENCODING	16 kbps CVSD	7.7 dB INTELLIGIBILITY MARGIN BELOW DESIGN POINT
DIGITAL MODULATION	BIPHASE PSK	VOICE-ACTIVATED CARRIERS
CARRIER-TO-NOISE DENSITY	51.3 dB-Hz	10^{-3} BER, 2.5 dB LOSS
LOW-RATE DATA	VARIABLE TO 16 kbps	MICROPROCESSOR CONTROLLED
DAMA NETWORK	CENTRALIZED	MICROPROCESSOR CONTROLLED
• SIGNALLING	16 kbps	SHARED MODEM W VOICE/DATA
• CALL REQUEST	1 CHANNEL/BAND	RANDOM ACCESS, UNSLOTTED
• MCS ASSIGNMENT	1 CHANNEL/BAND	COMMON MODE BROADCAST
MOBILE TERMINAL		
• RF POWER	40 WATTS	
• RECEIVER G/T	-26 dB/^{o}K	EDGE OF COVERAGE
• ANTENNAS	OMNIDIRECTIONAL-HEMISPHERIC BENT MONOPOLE-TURNSTILE	UHF RECEIVE, L-BAND TRANSMIT
SPACECRAFT EIRP	50 dBW/SPOT BEAM	3 dB BACKOFF, 4.5 dB VOX
SPACECRAFT G/T	3 dB/^{o}K	
MOBILE TERMINAL TRAFFIC	1125/SPOT BEAM	FULL DUPLEX, 1% BLOCKING

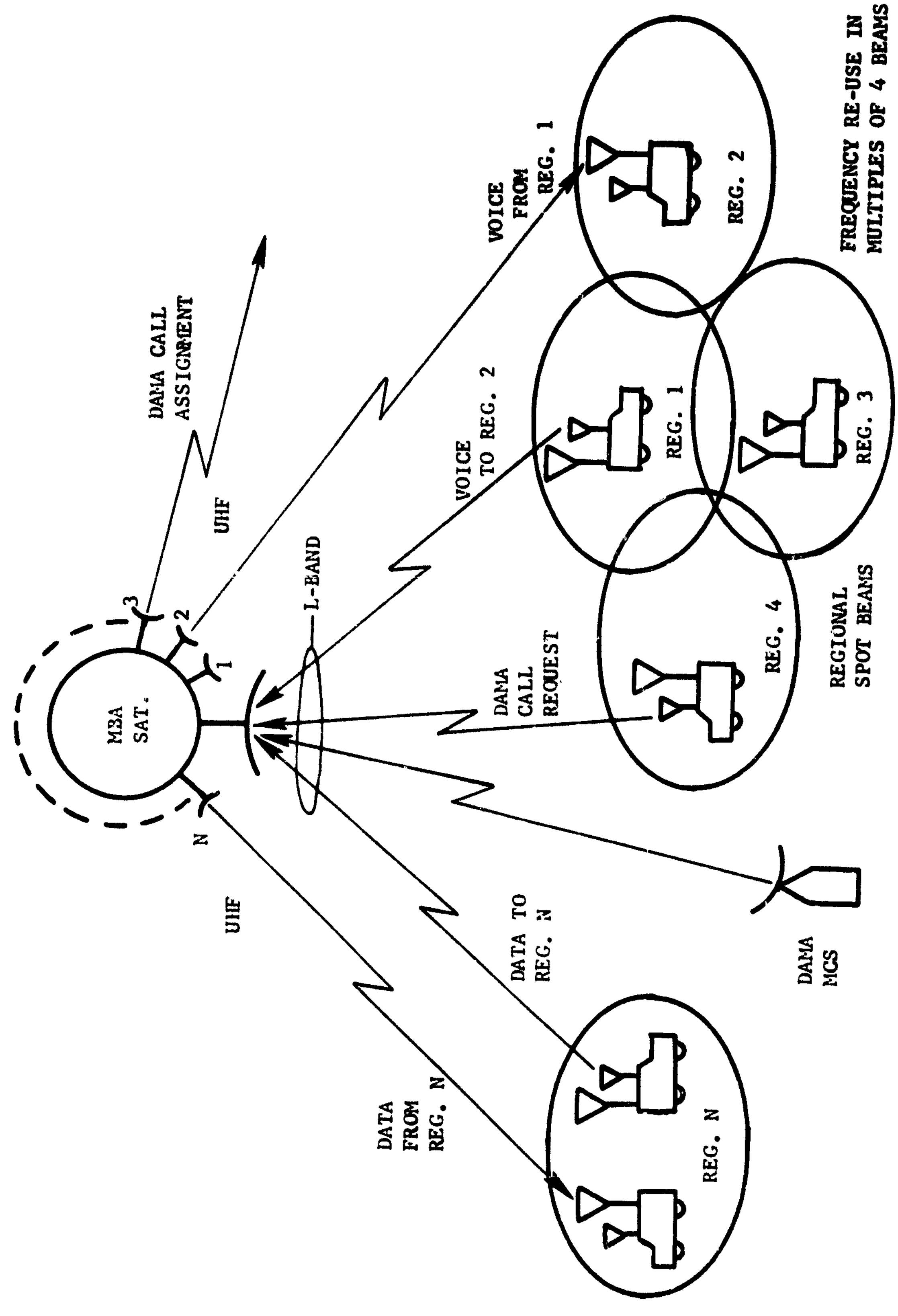


Figure 8.0-1. Conceptualized Operational System

channel assignment, with all channels assigned to a specific frequency and spaced with a fixed frequency spacing between channels.

8.1.1 Traffic/Frequency Plan. A proposed configuration for the satellite design of the operational mobile radio network is shown in Figure 8.1-1. Each mobile terminal in a given service region would transmit at L-band to a single common*, global receive antenna on the satellite. The satellite would filter and translate the N up-link frequency bands to N separate down-link spot beams in the UHF mobile band that are uniquely dedicated to the N service regions. This use of dual frequency bands for transmit and receive is forced by the lack of sufficient frequency space at UHF to accommodate a large number, N, of service regions.

Figure 8.1-2 illustrates a proposed frequency plan allocation for the operational system. Each service region would be assigned a specific frequency band on the up-link at L-band. The individual frequency bands would be wide enough to accommodate 56 channels, two of which would be reserved for DAMA control. Within each frequency band the 56 channels are spaced by 45 kHz, a channel spacing commonly used for SCPC systems. BPSK modulation at 16 kbps is employed to achieve the goal of a low-cost user terminal. (QPSK modulation offers the advantage of requiring half the bandwidth of BPSK but at the expense of a higher implementation margin and increased modem complexity which translates into higher cost.)

The UHF down-link frequency bands would be unique only to the adjacent service regions. Thus, if service Region 5 were bounded by Regions 3, 4, and 6, then Region 5 would require a unique frequency band assignment apart from Regions 3, 4, and 6. However, the frequency band assigned to Region 5 may be used over again for any region not adjacent to Region 5 nor adjacent to another region that used the same frequency band.

Operationally, if a mobile terminal in Region 1 wanted to communicate with a mobile terminal in Region 5, then the Region 1 terminal would tune its L-band transmitter to the appropriate DAMA-assigned channel in the frequency band reserved for Region 5. Similarly, the Region 5 terminal would tune its

*In practice the common global receive antenna on the satellite might be realized as, say 4 separate antennas jointly covering CONUS to provide additional gain at L-band. This approach tends to relieve the need for high values of satellite G/T (+3 dB/°K employed here).

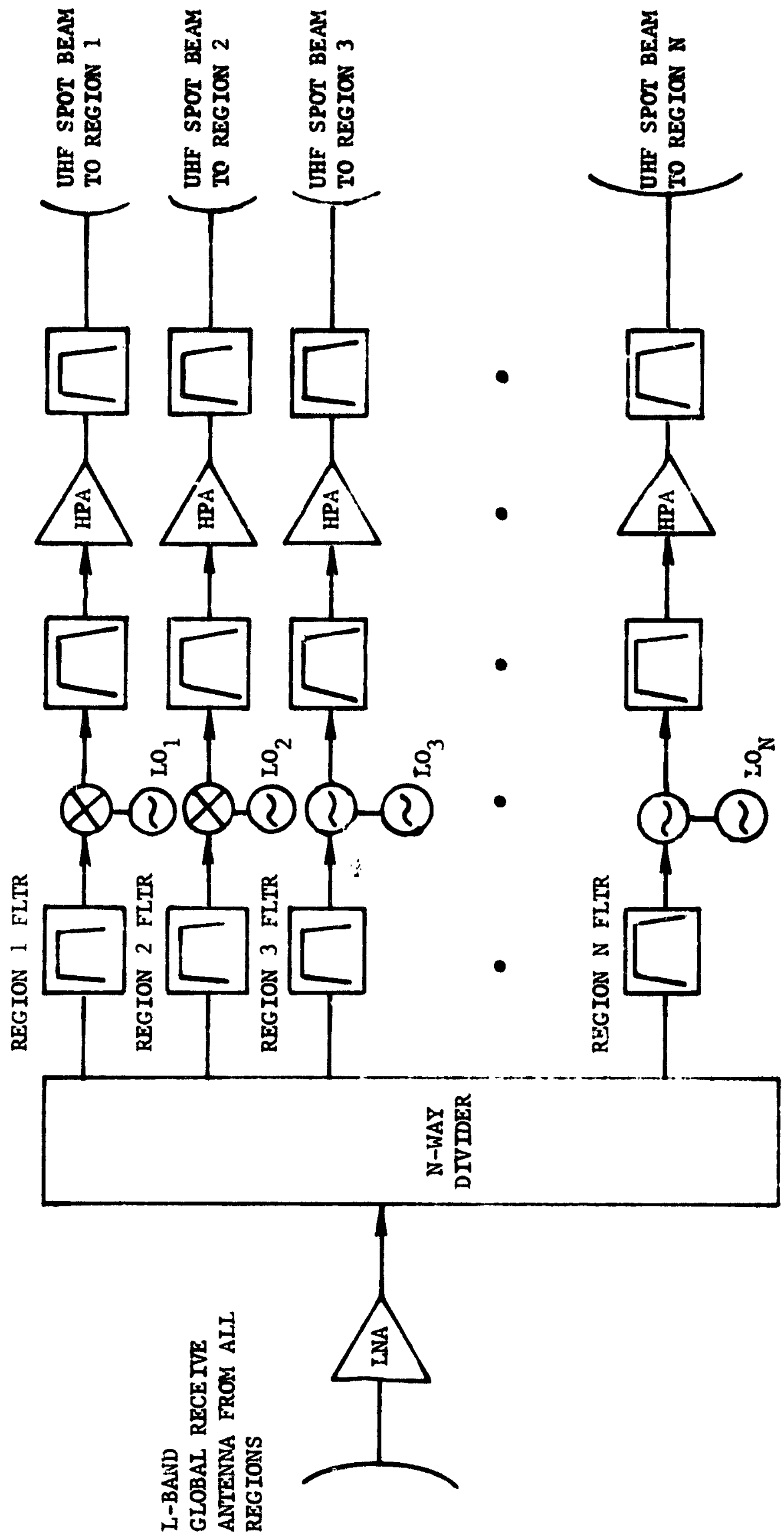


Figure 8.1-1. Mobile Radio Satellite Configuration

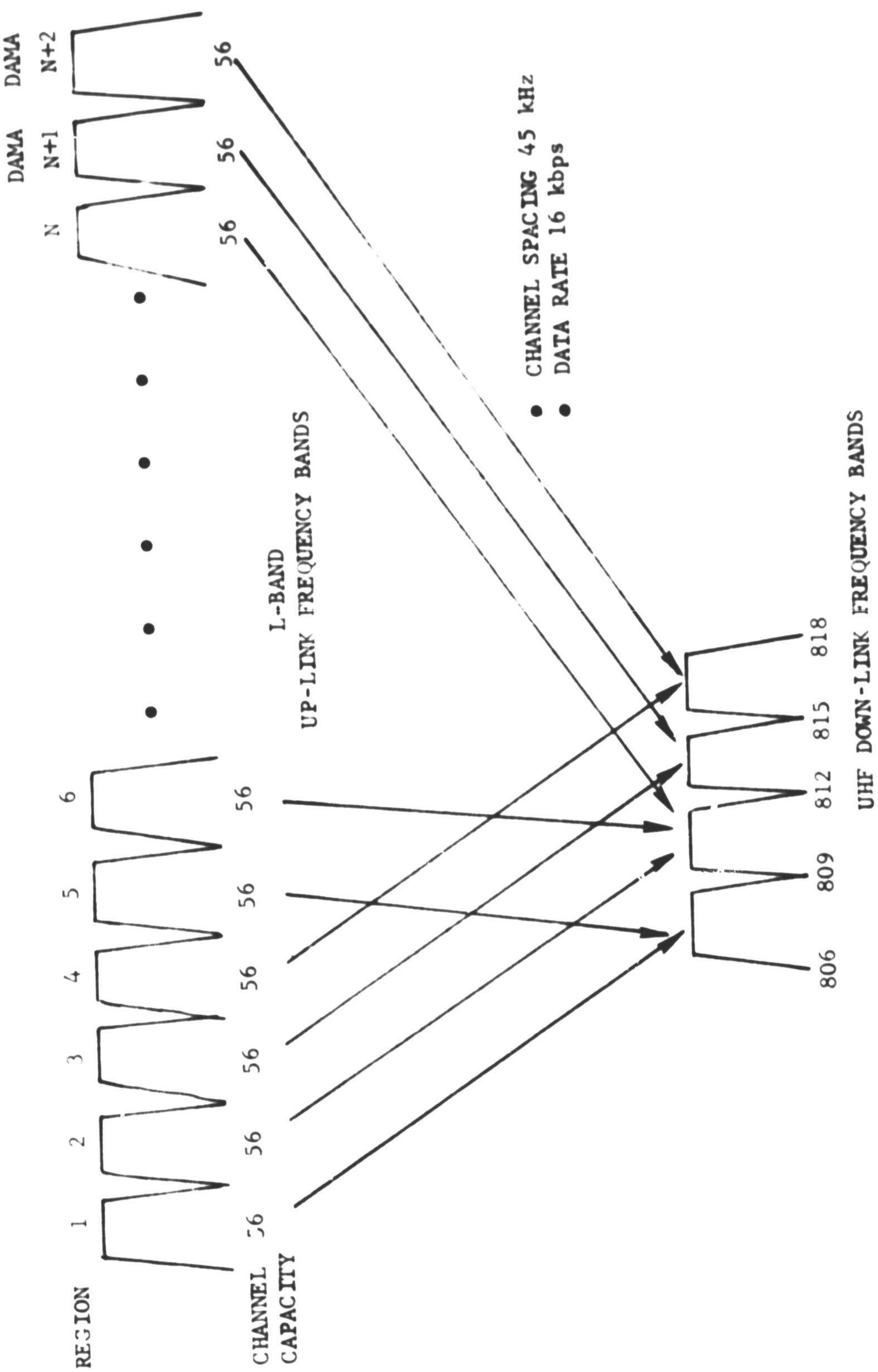


Figure 8.1-2. **Operational System Frequency Plan**

UHF receiver to the same DAMA-assigned channel. Conversely, for the return link the Region 5 terminal would tune its L-band transmitter to the DAMA-assigned channel in the frequency band reserved for Region 1, and the Region 1 terminal would tune its UHF receiver to the same DAMA-assigned channel. Obviously, to be viable this type of frequency plan requires the coordination and supervision of a suitable DAMA system for channel requests and assignments.

Random access among the ground terminals would be accomplished by a demand assignment multiple access system. The DAMA system implementation would require a central control terminal that determined all frequency assignments by a master computer. Each user ground terminal would require a microprocessor for station control. Communications between the master computer and the terminal microprocessor would be accomplished by sharing the user terminal digital communications channel between the DAMA mode and the voice mode.

Use of this type of frequency plan requires a L-band transmitter that tunes over a very wide frequency range. For N service regions the required transmit bandwidth is (N+2) x 56 x (.045 MHz) + (N+3) x (0.5 MHz), assuming guard bands of 0.5 MHz between each regional band and two spot beams allocated for DAMA. For example, to provide for 40 service regions each supporting 56 channels, the required total up-link bandwidth is 127.34 MHz. Since such wide bandwidth allocations are not available at UHF, this plan forces the postulation of a L-band allocation.

However, the UHF receiver at each terminal would have to tune over only a very limited range of 55 x 0.045 MHz = 2.475 MHz, with the overall receive band being about 45 kHz wider for modulation sidebands (i.e., 2.52 MHz wide) in the 806 to 890 MHz band. Thus, each service region would require 3.02 MHz (2.52 + 0.5) of frequency space in the UHF mobile communications band. This bandwidth includes an allocation of 0.5 MHz for guard bands.

Obviously, the required up-link bandwidth is very wide depending upon the number of regions served. This wide bandwidth could be reduced by various schemes such as lowering the amount of guard band between frequency bands (at increased satellite filter complexity), using QPSK modulation rather than BPSK, and/or use of spectral shaping of the transmit signal (at greatly increased terminal costs and complexity). Another bandwidth reduction scheme is to use

different polarizations as well as different frequencies for each up-link, assuming at least 30 dB isolation could be obtained technically and economically between two input frequencies at opposite polarizations. Unfortunately, all of these techniques impact the system design in terms of increased complexity and costs thereby defeating the study objectives and therefore are unacceptable.*

8.1.2 Ground Terminal Configuration and Operation. A block diagram of the mobile radio terminal is shown in Figure 8.1-3. The telephone handset would be a four wire telephone, to preclude the need for echo suppressor circuits, with a touch pad for dialing and an audible SONALERT unit for ringing. Echo suppressor and telephone line conditioning circuitry could be included where it may be required to interface with conventional two-wire telephone circuits.

A typical call sequence diagram for the DAMA operation is illustrated in Figure 8.1-4. Although the following discussion is only a brief outline of terminal operation, intended to give an overview of system operation, an in-depth description of a DAMA call sequence referring to Figure 8.1-4 is given later in Section 8.5.1.

To initiate a call in the DAMA mode the calling user would take his handset off-hook, thus sending an off-hook indication to the Digital Control Unit (DCU). The DCU would then send dial tone (via a data format to the delta mod decoder) to the calling user indicating a ready request for service. The user would then punch in, via the touch pad, the called subscriber number. Audible tone feedback would be supplied by the DCU to the calling user as each number was punched in. After collecting all of the numbers, the DCU would scan them for obvious errors (invalid number) and then transmit a request for circuit assignment plus the called number to the central control computer at the Master Control Station (MCS) via a random access common signalling data channel (CSC) at a 16 kbps rate. The central control computer would receive the request and number and determine if the called terminal was busy or not (by a memory search since the central control computer knows the status of all DAMA terminals at all times). If the called terminal is busy or unavailable for any reason the central control computer informs the calling terminal via a broadcast common signalling (BCC) data channel that the call cannot be completed. The calling terminal DCU then sends a busy tone to the user handset and will no longer accept any new calls until the user goes on-hook and then off-hook again.

*However, regardless of the scheme used, in order to avoid interference problems with terrestrial systems the entire up-link frequency band and the down-link frequency band for each region must be reserved for this usage.

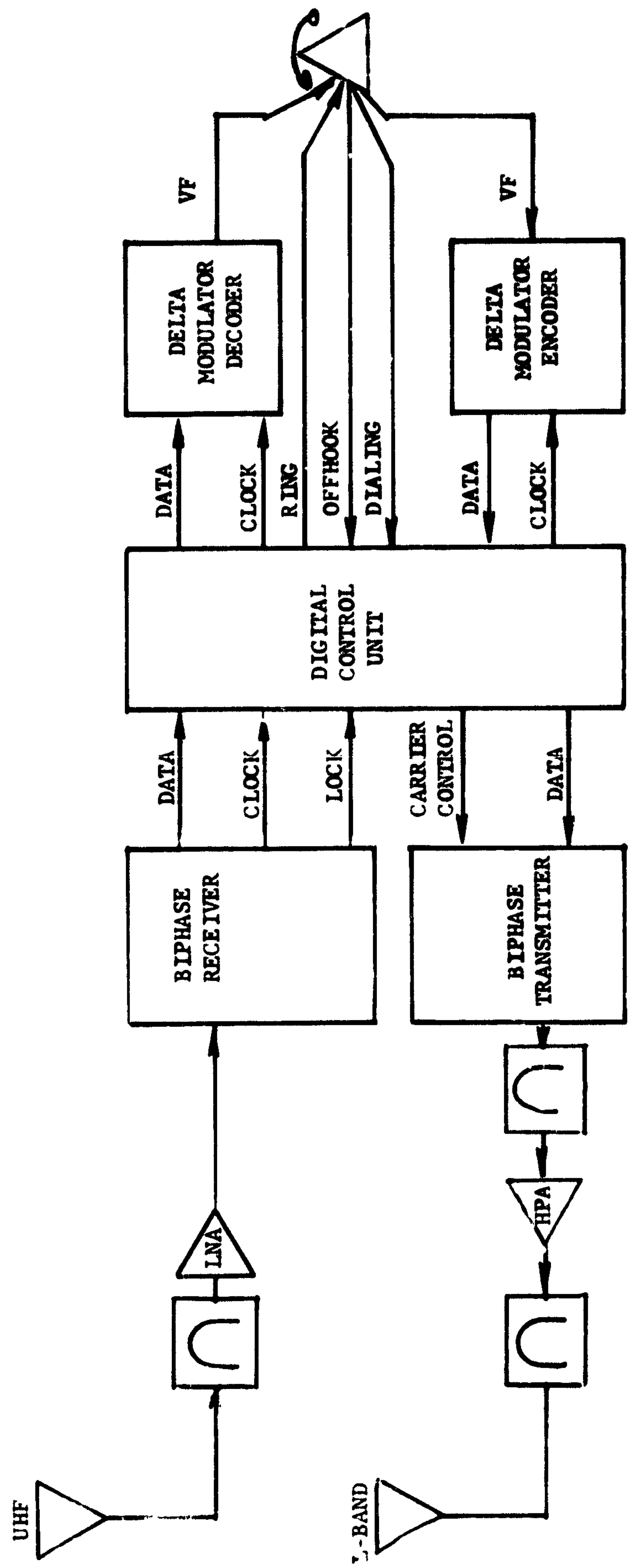


Figure 8.1-3. Mobile Radio Terminal Configuration

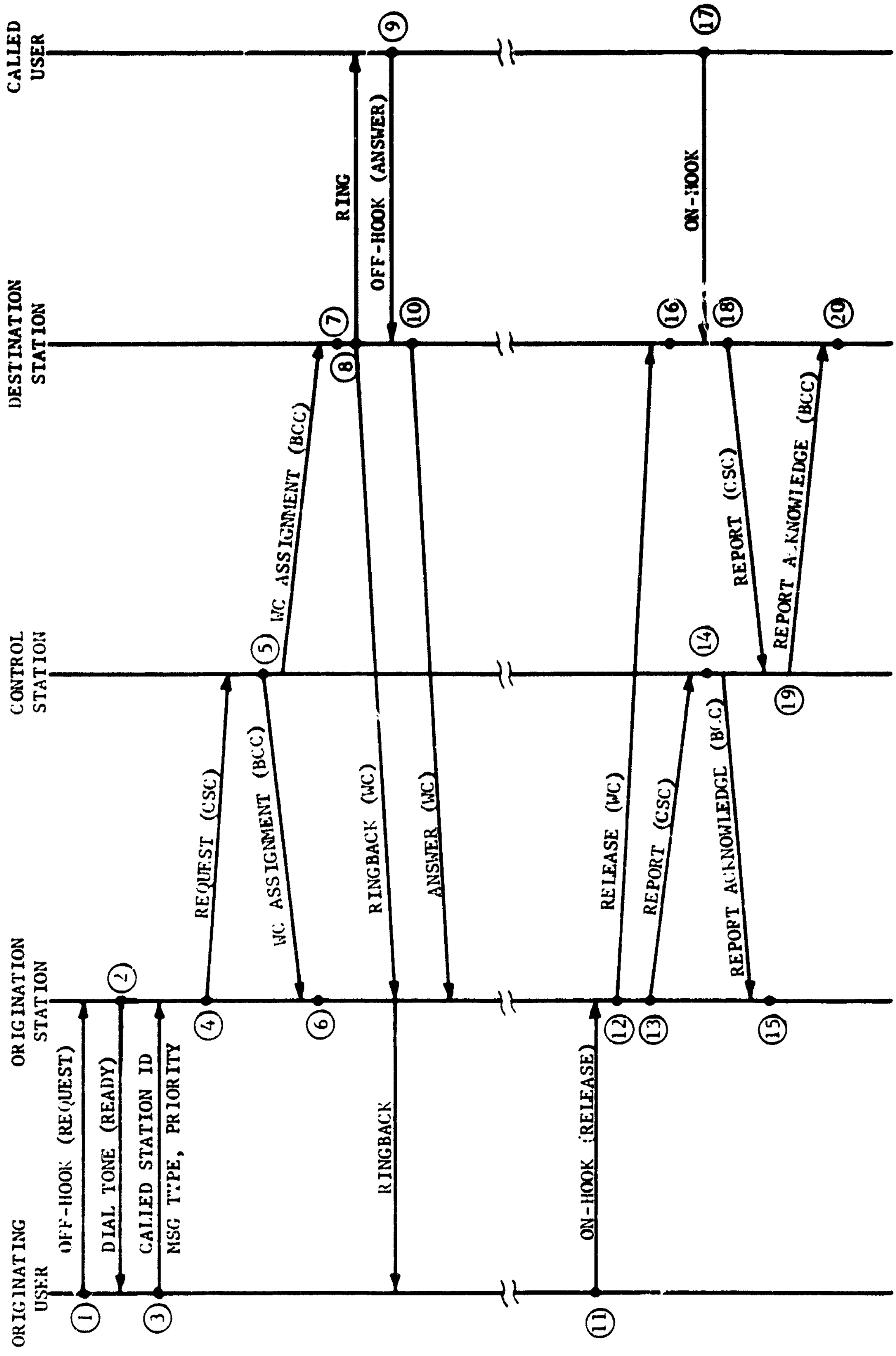


Figure 8.1-4. DAMA Call Sequence

However, if the called terminal is available, the central control computer sends frequency assignments to both the calling and called terminals. The called terminal DCU then sends interrupted ring signals to its user telephone and ringback tone (via the 16 kbps data stream) to the calling terminal for positive feedback that the called party is being rung. When the called party answers the telephone (goes off-hook) the ring tone stops along with the ringback tone and a through path is established for the call.

The two mobile terminals then talk to each other using voice-spurt-activated carrier operation (VOX) where a carrier is transmitted only when the user is talking. In order to prevent first syllable clipping due to finite VOX operation times and receiver lock-up times, digital buffer lines are used. During quiet periods when a terminal is not receiving any carrier, the DCU sends an idle pattern (1010 pattern) to the delta modulation decoder circuit for a squelch condition rather than allow a random pattern or steady state pattern to go into the decoder.

The call continues in the conversation mode until either terminal goes on-hook. At this time the DCU notes the on-hook condition, retunes the modem to the common signalling data channel frequency, and reports to the central control computer that it is idle now and available for calls. The MCS acknowledges back via the BCC that the terminal is now idle. The network is now ready to process a new calling sequence.

In summary the calling sequence procedures described here are intended to be transparent to both the calling and the called parties and to imitate a standard land-line telephone network. Since these procedures are familiar to the terminal users, it makes the mobile system readily acceptable to even untrained users.

8.2 Mobile Terminal Design Description

In the earlier sections that considered the various theoretical trade-offs associated with voice digitization and digital modulation techniques it was concluded that CVSD at 16 kbps with BPSK is the preferred combination of techniques to select for the digital mobile application. This conclusion is employed here

with one minor modification; the assumed implementation loss. For theoretical considerations the implementation loss was taken to be 1 dB. However, for an operational system a larger value is indicated since the resulting cost of the user terminal is proportional to how close to the theoretical performance the actual terminal performance approaches. The use of loose specifications on component tolerances and performance factors and the use of simple, low-cost circuit techniques versus more complex and expensive techniques, means overall lower costs but higher departure from theoretical performance.

In order to keep the mobile terminal costs as low as possible, as would be essential with a terminal population of up to 10,000 or more, adoption of a high implementation margin is necessary. Therefore, the operational terminal design here is based on a 16 kbps CVSD codec with BPSK modulation allowing the performance deviation from theoretical (implementation margin) to be as great as 2.5 dB. Thus, the total required C/kT is 51.3 dB-Hz (48.8 + 2.5). This assumption reduces the design point link margin to only 0.5 dB. Table 8.2-1 summarizes the operational system link budget and link margin. However, the expected performance of the terminal at the worst antenna look angle on both satellite and user terminal simultaneously would be a BER of better than 10^{-3}. The signal level from the expected design operating point could degrade an additional 3.0 dB before the threshold BER of 10^{-2} was reached. A total overall degradation of 8.2 dB could occur before reaching the usable end point BER of 10^{-1} at which point CVSD is still intelligible but low in quality. It should be pointed out that the full 8.2 dB margin may not be achieved in practice because of design tradeoffs as discussed later in the Hardware Considerations section under carrier acquisition. Perhaps, the total degradation achievable for the worst antenna look angles would be of the order of 5 to 6 dB.

Figure 8.2-1, Mobile Terminal Configuration, describes the major subsystems within the ground terminal. The transmit signal path starts with the user handset and goes to the delta modulator coder unit, CVSD, where the voice is converted into a 16 kbps digital data stream. From the codec the digitized voice output goes to the Digital Control Unit (DCU) where the data stream is processed so that the transmit carrier is turned on only when speech is present and buffered to prevent first syllable clipping. The DCU outputs a frequency control assignment to the transmit frequency synthesizer (as per remote DAMA command or local manual control. This DCU unit not only selects the channel assignment for the called

C-2

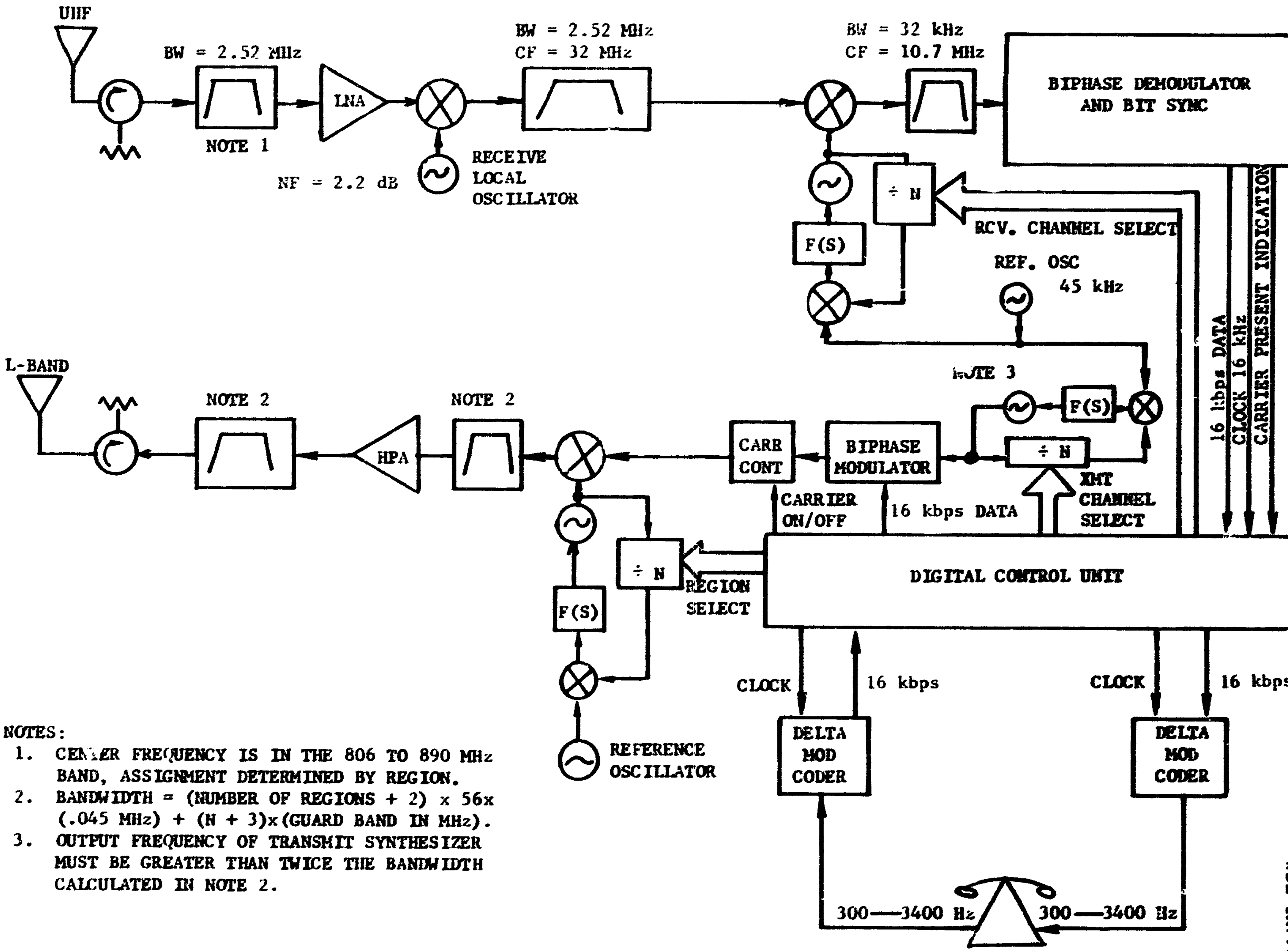
Table 8.2-1. Operational System Link Budget and Margin

LINK BUDGET

UP-LINK FREQUENCY	1650 MHz
MOBILE TERMINAL EIRP	16.0 dBW
UP-LINK PATH LOSS	-189.2 dB
BOLTZMAN'S CONSTANT	228.6 dBW/^{o}K-Hz
SATELLITE G/T	3.0 dB/^{o}K
UP-LINK G/T	58.4 dB-Hz
DOWN-LINK FREQUENCY	868 MHz
SATELLITE EIRP	50.0 dBW
TWTA BACKOFF	-3.0 dB
VOX OPERATION	4.5 dB
DOWN-LINK PATH LOSS	-183.6 dB
BOLTZMAN'S CONSTANT	228.6 dBW/^{o}K-Hz
MOBILE TERMINAL G/T	- 26.0 dB/^{o}K
56 CHANNELS PER SPOT BEAM	- 17.5 dB
DOWN-LINK C/kT PER CHANNEL	53.0 dB-Hz
COMBINED C/kT PER CHANNEL	51.8 dB-Hz

LINK MARGIN

MODULATION TECHNIQUE	BPSK
VOICE DIGITIZATION TECHNIQUE	CVSD
16 kbps DATA RATE	42.0 dB-Hz
DESIGN POINT E_b/N_o (10^{-3} BER)	6.8 dB
REQUIRED THEORETICAL C/kT	48.8 dB-Hz
IMPLEMENTATION LOSS	2.5 dB
REQUIRED PRACTICAL C/kT	51.3 dB-Hz
AVAILABLE LINK C/kT	51.8 dB-Hz
DESIGN POINT LINK MARGIN	0.5 dB
ADD'L. AT THRESHOLD (10^{-2} BER)	2.5 dB
THRESHOLD LINK MARGIN	3.0 dB
ADD'L. AT USABLE END POINT (10^{-1} BER)	5.2 dB
USABLE END POINT LINK MARGIN	8.2 dB



NOTES:
1. CENTER FREQUENCY IS IN THE 806 TO 890 MHz BAND, ASSIGNMENT DETERMINED BY REGION.
2. BANDWIDTH = (NUMBER OF REGIONS + 2) x 56x (.045 MHz) + (N + 3)x(GUARD BAND IN MHz).
3. OUTPUT FREQUENCY OF TRANSMIT SYNTHESIZER MUST BE GREATER THAN TWICE THE BANDWIDTH CALCULATED IN NOTE 2.

Figure 8.2-1. Mobile Terminal Configuration

region, but outputs a 16 kbps data stream to the biphase modulator, and sends an on/off control signal to the carrier control unit. The biphase modulated carrier passes through the carrier control unit to the up-converter mixer/filter which passes only the upper sideband signal generated by mixing of the local oscillator and the synthesizer carrier signals. Tuning of the local oscillator is controlled by the DCU to transmit a carrier at the called terminal region's center frequency. The upper sideband signal then goes to the High Power Amplifier (HPA) where the signal level is raised to about 40 watts of output at the transmit antenna and up to the satellite at L-band.

The UHF receive signal from the satellite is picked up by the mobile terminal's receive antenna and routed to the receive filter and low noise amplifier (LNA). This receive filter must have sufficient selectivity to reject the transmit signal from the HPA to prevent overloading of the LNA and generation of high receive IM levels. The LNA output goes to a mixer/filter combination that down-converts the receive signal to a 30 to 34 MHz band. A receive frequency synthesizer, contolled by the DCU, picks out individual channels in the receive band by mixing only the desired carrier down to a 10.7 MHz center frequency. The biphase demodulator accepts receive carriers at 10.7 MHz and recovers data and clock from the demodulated signal and transmits them to the DCU. In addition, the demodulator also provides a logic signal to the DCU indicating whether a receive carrier is present or not. Processing of the receive data stream is performed by the DCU to prevent last syllable clipping when the receive carrier disappears and the receiver must be squelched (refer to Receiver VOX Operation Section for details). The DCU then routes the processed data to the delta modulator decoder where it is converted back to an analog voice signal and sent to the user's handset.

8.3 Hardware Considerations

This section deals with the system and circuit design factors and parameters that impact the hardware design of a mobile terminal meeting the specification of Section 8.2. The primary consideration is minimizing the terminal hardware complexity to achieve a low per unit terminal cost and ensure high reliability. Numerous and judicial tradeoffs between performance and costs must be made to achieve an economical and viable system. For example, BER obviously can

be improved by lowering the noise temperature of the receiver front-end, by using a quieter front-end amplifier, but at an increased cost. Similarly, the terminal costs can be reduced by using a low-cost predetection filter that does not have tightly controlled parameters such as passband ripple, group delay, and skirt selectivity. However, the use of filters with relaxed specifications creates higher intersymbol interference and adjacent channel interference effects that can be expected to degrade the BER performance from that theoretically possible.

From a production point of view, costs can be controlled best where components do not need to be selected, hand assembly is low, test time is minimal, low-cost components are used throughout, and unskilled personnel can assemble and test the completed unit. These items are generally in direct conflict with achieving a low implementation margin, thus the reason for arguing for a high implementation margin. The adopted 2.5 dB implementation margin is considered to be a reasonable compromise between high quality technical performance and low cost.

8.3.1 Transmit VOX Operation. In order to improve satellite power utilization, it is desirable to have the ground terminal transmit a carrier only when the user is actually talking; namely a voice-operated carrier (VOX) system. Implementation of the VOX feature is relatively easy, since the normal transmit output of the delta modulator codec is a 1, 0, 1, 0 -- pattern during idle conditions (ignoring background noise), with an average value of 0.5. A simple VOX algorithm for speech presence detection is to digitally integrate the codec output over a moving block of data and activating the transmit carrier whenever the integrated value exceeds or drops below a fixed digital threshold sum. A more sophisticated algorithm should be used to reduce false carrier activation due to background noise and would be suited ideally for a microprocessor. However, a microprocessor would be cost effective only where it was required for DAMA operation but otherwise idle when a call was in progress. This is one of the functions performed by the DCU.

Since the VOX algorithm has to work over past data, the decision to turn on the transmit carrier occurs a finite time interval after speech has started. To prevent first syllable clipping the speech data must be buffered before being applied to the biphase modulator. The length of the buffer depends upon the length of the block of data over which the algorithm works plus processing time.

In order to achieve fast carrier acquisition and clock recovery in the receiver, the DCU should also add a preamble to the data applied to the modulator prior to the speech data pattern. For example, such a preamble might be the bit pattern with the greatest number of transitions per unit of time (for fast clock recovery), the 1010---bit pattern. However, a preamble pattern for resolving phase ambiguity in the receiver is not necessary for reasons discussed later in the demodulator section.

Figure 8.3-1, Transmit VOX Timing Diagram, details the sequence of events that occur when a speech signal is applied to the transmit codec. In this illustration the transmit signal is a representation of the analog signal from the user's handset showing the onset of speech. The codec output represents the digital data stream generated as a result of the transmit analog input signal. Also shown is the carrier on/off signal, the output of the VOX detector circuit that decides a speech signal has occurred and that the carrier should be turned on. The VOX decision delay interval is the time it takes from the onset of speech before the VOX detector circuit reacts to turn on the carrier. This delay interval typically would lie in the one to four millisecond range (160 to 640 bits at 16 kbps). A long delay interval permits more decision time to reduce false carrier activations due to noise. The preamble interval is the time required by the receiver to lock-up on the carrier, recover both clock and data, and deactivate the receiver squelch. A typical preamble time would be one to six milliseconds, depending upon the receive E_b/N_o value and other factors. The sum total of the VOX decision delay interval and the preamble interval is the data buffer interval. This is the amount of data that must be buffered to prevent loss of the data in the transmitter or receiver circuits due to finite decision and lock-up times. Without sufficient data buffering, first syllable clipping of the voice signal will occur at the start of every transmission.

8.3.2 Transmit Modulation Sidebands. The power spectral density for ideal BPSK with rectangular waveforms extends to infinity on both sides of the carrier frequency in a $(\sin X)^2/X^2$ distribution. Since each transmitted carrier is assigned a fixed bandwidth, any transmitted energy that falls outside the allocated channel bandwidth causes interference in the adjacent channels. Due to the fall off of the energy away from the carrier, the most significant interference occurs in the immediately

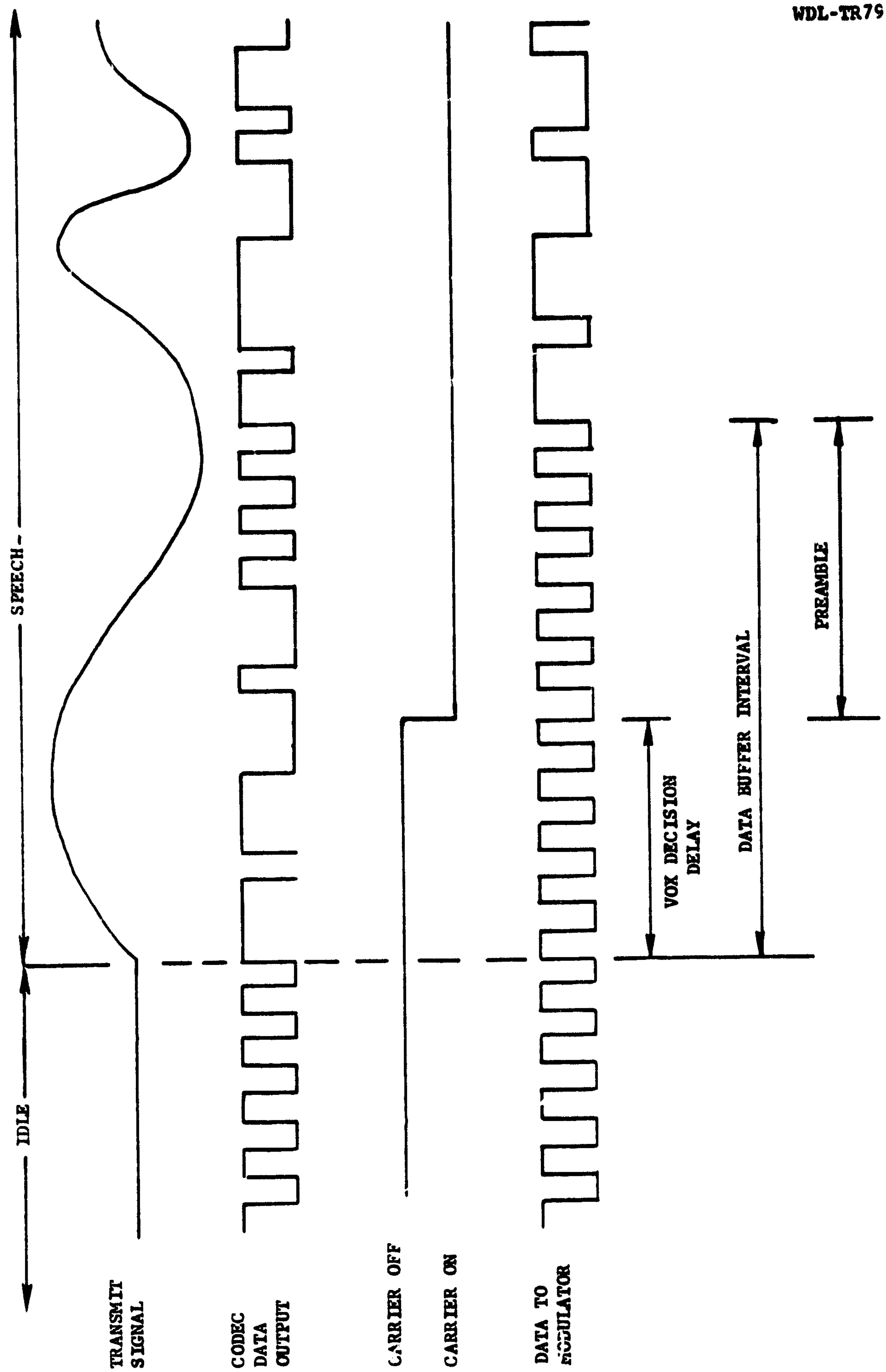


Figure 8.3-1. Transmit VOX Timing Diagram

adjacent SCPC channels on either side of the transmit channel. The objective of the discussion here is to point out the potential problem and to suggest possible solutions.

Significant reduction of the sideband energy falling in adjacent channels can be achieved by shaping the data stream in the time domain. A relatively simple circuit to shape the data stream can be designed that generates a transmitted spectrun where adjacent channel interference beyond 16 kHz on either side of the carrier is negligible. Thus, a closer channel spacing at less than, say, twice the data rate could be accomplished.

Spectral shaping, however, requires linear modulation and linear power amplification in the signal path through the transmitter, satellite, and receiver circuits through the predetection filter. Nonlinear operation at any point in the link removes or distorts the AM components created by data shaping and effectively restores the $(\sin X)^2/X^2$ power spectral distribution.

Due to the necessity to operate the satellite transponder and terminal receiver circuits linearly to minimize IM distortion products arising from the large number of SCPC carriers in the frequency band, the satellite transponder and terminal receiver circuits must be linear with or without spectral shaping. However, it is very costly to make the mobile terminal transmitter output amplifier linear. Not only must the power capacity of the output amplifier be greater than that actually transmitted, but lower amplifier efficiencies will occur that increase the power consumption of the terminal, which is undesirable for mobile applications.

In addition, spectral shaping increases intersymbol interference effects which can lead to significant system degradation unless complex equalizer circuits are used in the receiver.

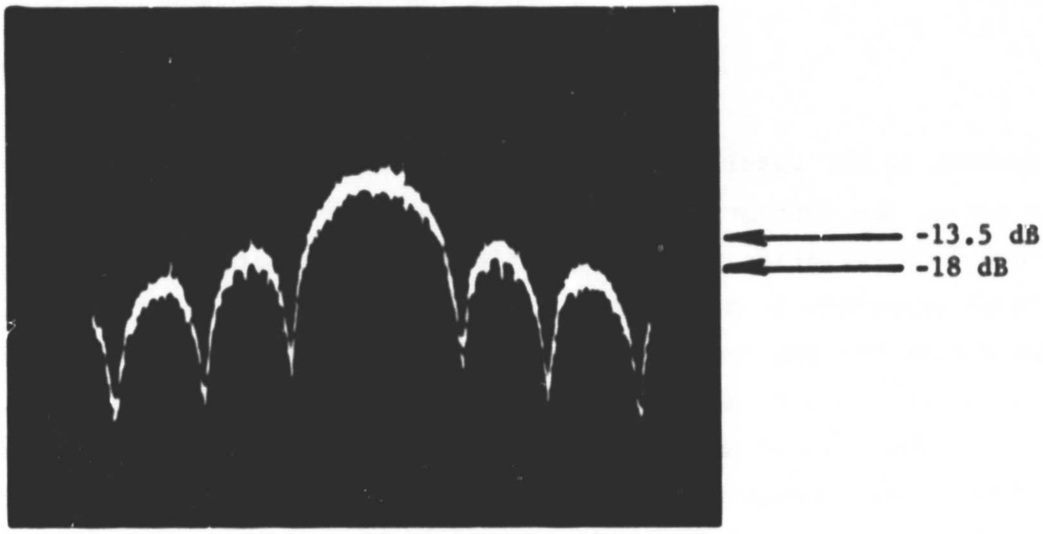
However, the most compelling reason for spectral shaping other than bandwidth conservation is due to the possibility of unequal transmit levels arriving at the satellite from each terminal. These unequal power levels are due to other terminals transmitting at different elevation/azimuth points on the terminal's antenna pattern, variations in absolute transmit output level due to hardware

plus variations in the satellite global receive antenna pattern. Assuming a 3 dB variation due to the antenna transmit pattern, a 1 dB variation in terminal output power, and a 3 dB variation in satellite global receive pattern, a worst case channel level variation from different terminals could be as much as 7 dB. Figure 8.3-2A shows that the peak of the second sideband lobe for unshaped BPSK is typically 18 dB below the peak of the main lobe. Hence, for a channel spacing of this order the adjacent channel main lobe could possibly be only 11 dB above the second sideband lobe of the center channel producing a mutual interference effect. With a 45 kHz channel spacing, as employed here, the adjacent channel interference level is on the order of 19 to 20 dB down resulting in a worst case level of 12 to 13 dB, not much better than the example case.

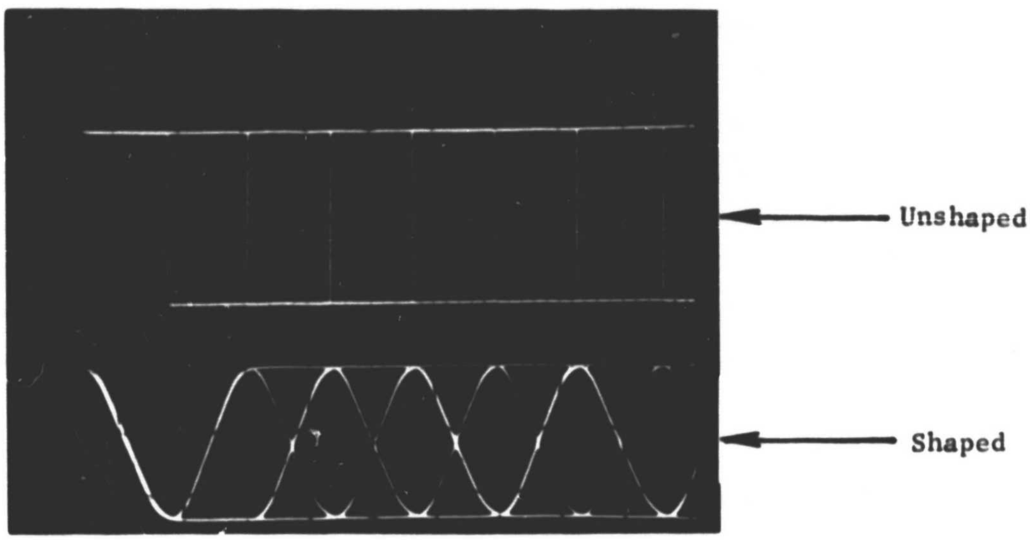
Figure 8.3-2C, shows the output spectrum of data that has been shaped by a simple shaping circuit to generate a raised cosine modulation pattern for every transmitted bit, as shown in the bottom trace of Figure 8.3-2B. The output spectrum clearly shows sideband levels 45 kHz away from the carrier that are on the order of 40 dB below the main lobe, thus little degradation of the adjacent channel should occur. This particular shaped-data pattern shown here is not necessarily the desired one, but it does demonstrate the value of data shaping.

Due to the additional cost and complexity of spectral shaping that would be added to the mobile terminal, a simpler approach is to use a wider channel allocation (45 kHz versus less than or equal to 32 kHz) at the expense of bandwidth efficiency. Although this is the approach chosen at this time, future system considerations and circuit implementations may make spectral shaping a viable alternative or even a necessity. When spectral shaping is used, the data detection filter must be modified to achieve a closer matching with the shaped data.

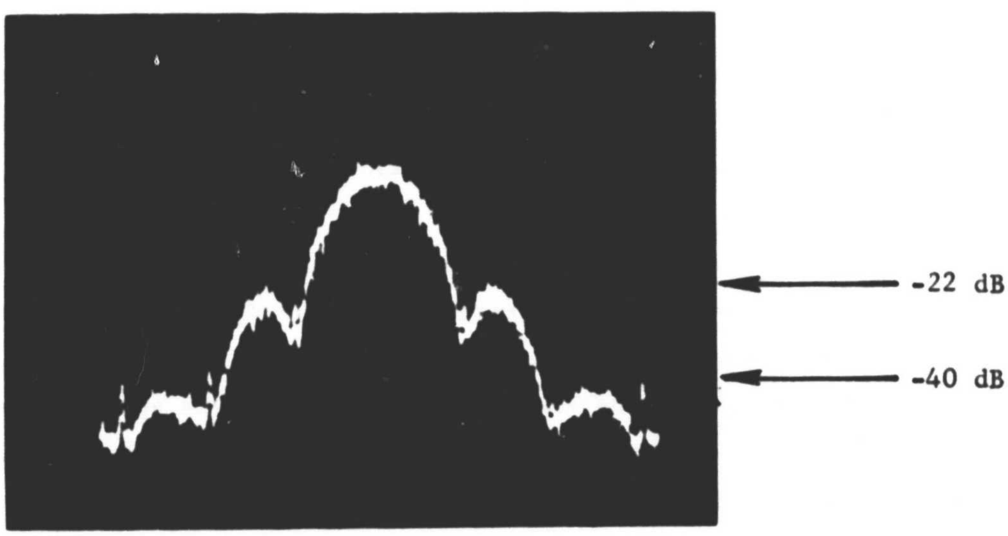
8.3.3 Receiver Carrier Acquisition. VOX operation requires a rapid lock-up time (<4 ms typical) in the receiver carrier reconstruction loop and bit synchronizer circuits at the lowest E_b/N_o operating point. Rapid acquisition dictates that the loop bandwidth of the carrier reconstruction circuit should be approximately equal to the worst case frequency offset of the incoming carrier from the rest frequency of the carrier reconstruction loop VCO.



A: Unshaped Data Power Spectrum



B: Data Waveforms W/WO Shaping



C: Shaped Data Power Spectrum

Figure 8.3-2. Measured Power Spectra - 16 kbps BPSK W/WO Data Shaping

However, the carrier reconstruction loop bandwidth also must be as narrow as possible in order to maximize the signal to noise ratio in the carrier reconstruction loop when operating at the lowest E_b/N_o values. If the SNR is not kept high enough, then the BER performance is degraded and, perhaps even worse, cycle slipping will occur causing polarity reversals in the data stream. Cycle slipping would lead to more serious voice distortion effects than single errors in the bit stream. Actual tests of the proposed carrier reconstruction circuit indicates that it will maintain lock with low phase jitter with a SNR in the loop of at least 13 dB and avoid cycle slipping. However, further analysis and evaluation is required to determine the minimum SNR value necessary since the magnitude of these effects are unknown.

The overall carrier frequency offset appearing at the input to the data demodulator is primarily the sum of the frequency error of the transmitted signal, the frequency error of the satellite translation oscillator, the doppler frequency shift from the satellite, and the frequency error of the mobile terminal down-converter local oscillator. In addition to these factors, in the absence of an input carrier, the demodulator VCO will assume the average frequency of the input noise spectrum. This is true, ignoring the effects of circuit bias offsets and any input spurious signals. Since the predetection filter determines the noise spectrum that the carrier acquisition VCO sees, and this filter could have a center frequency offset of ±2 kHz, this amount must be added to the total frequency error at the demodulator.

Table 8.3-1 summarizes the overall contributors of the system frequency errors. Column A represents the worst case errors due to use of low-cost components in the terminals, whereas column B corresponds to the worst case errors expected for premium components. Column A would require a carrier acquisition loop bandwidth on the order of 5 kHz. Such a wide loop would virtually preclude operation at the low E_b/N_o values associated with the mobile terminal, since a high enough SNR could not be maintained in the loop. On the other hand, the requirements implied by Column B are reasonable and should allow operation down to an E_b/N_o of about 4 dB. Operation below this point is difficult to predict since the effects of VCO cycle slipping have not been verified and can only be estimated.

Table 8.3-1. System Frequency Errors

	"A" LOW COST COMPONENTS	"B" PREMIUM COMPONENTS
TRANSMITTED FREQUENCY ERROR	≤ 850 Hz	≤ 85 Hz
SATELLITE TRANSLATION ERROR	≤ 100 Hz	≤ 100 Hz
SATELLITE DOPPLER SHIFT	≤ 200 Hz	≤ 200 Hz
RECEIVER DOWN-CONVERTER ERROR	≤ 850 Hz	≤ 85 Hz
CARRIER ACQUISITION VCO ERROR	≤ 3000 Hz	≤ 1000 Hz
TOTAL (WORST CASE ERROR)	≤ 5000 Hz	≤ 1470 Hz

While the column A components appear to be insufficient to achieve successful operation, it is still possible to use a long loop AFC to correct for all system errors except for the transmitter frequency error and the carrier acquisition VCO offset error. Successful operation of the long loop AFC circuit would require the mobile receiver to always tune to a pilot carrier when the terminal is idle. Additional requirements are that the VCO have a low offset error and that system frequency errors have low drift over the longest carrier-off period expected during terminal operation.

A digital "bang-bang" servo loop would be the easiest circuit to implement for the long loop AFC. By using an up-down counter/digital-to-analog converter controlled by the AFC detector the carrier presence detector could stop the up-down counter operation in the absence of carrier, thus providing AFC memory when the carrier is off.

Because the long loop AFC circuit still requires a low VCO offset and adds additional circuitry, the costs are likely to be the same order of magnitude as using premium components. Further system definition is required to estimate the magnitude of the problem in any event. Certainly VOX operation is not a trivial item and greatly complicates the terminal design. Consequently, the final decision as to how to handle the carrier acquisition problem must be put off awaiting further analysis and experimenting.

8.3.4 Data Demodulator and Bit Synchronizer. The proposed configuration for the data demodulator is shown in Figure 8.3-3. This design is a digital implementation of the more classical linear Costas PLL and has the distinct advantage of being relatively inexpensive to implement compared to the conventional analog version, plus lending itself to LSI implementation in high volume production.

A unique feature of this data demodulator is that the carrier reconstruction loop hardware realization is achieved by the utilization of Modulo-2 adders (exclusive OR logic gates) for the I & Q detectors as well as for the Costas loop third multiplier. This digital implementation requires use of hard limiting following the 10.7 MHz receive IF bandpass filtering process to establish logic level voltages for the I & Q detectors. (A degradation of about 1 dB in signal-to-noise ratio would be incurred by the IF limiting process. However, the simplicity and lower costs of the digital implementation compared to the conventional analog version argues in favor of accepting the small degradation).

Low pass filters and hard limiters (logic level outputs) follow the I & Q detectors. The hard limiter in the Q channel establishes near infinite gain on the phase error estimate in the carrier tracking PLL. The final modulo-2 adder removes the data sign term from the phase error estimate. A side effect is that the near infinite loop gain causes the phase lock loop to be unconditionally unstable. This instability causes a continuous limit cycle loop oscillation. However, this oscillation is controlled with a small amplitude and a high frequency (compared to the data rate). A beneficial effect of the oscillation process is to actually linearize the PLL. Long term (compared to the limit cycle period) phase errors are manifest in duty factor asymmetry in the time and cycle waveforms, thus providing average phase correction. Hence, in the long-term sense, the loop is stable. With noise present at the input, noise excursions through zero phase error capture the Q channel limiter and the limit cycle oscillation disappears.

Subjectively, the noise has the effect of causing the loop gain to be finite and stable. As the noise increases (signal-to-noise ratio decreases), two simulateous effects occur. The noise captures more and more of the IF limiter output power, causing the signal power to diminish, and the Q detector output

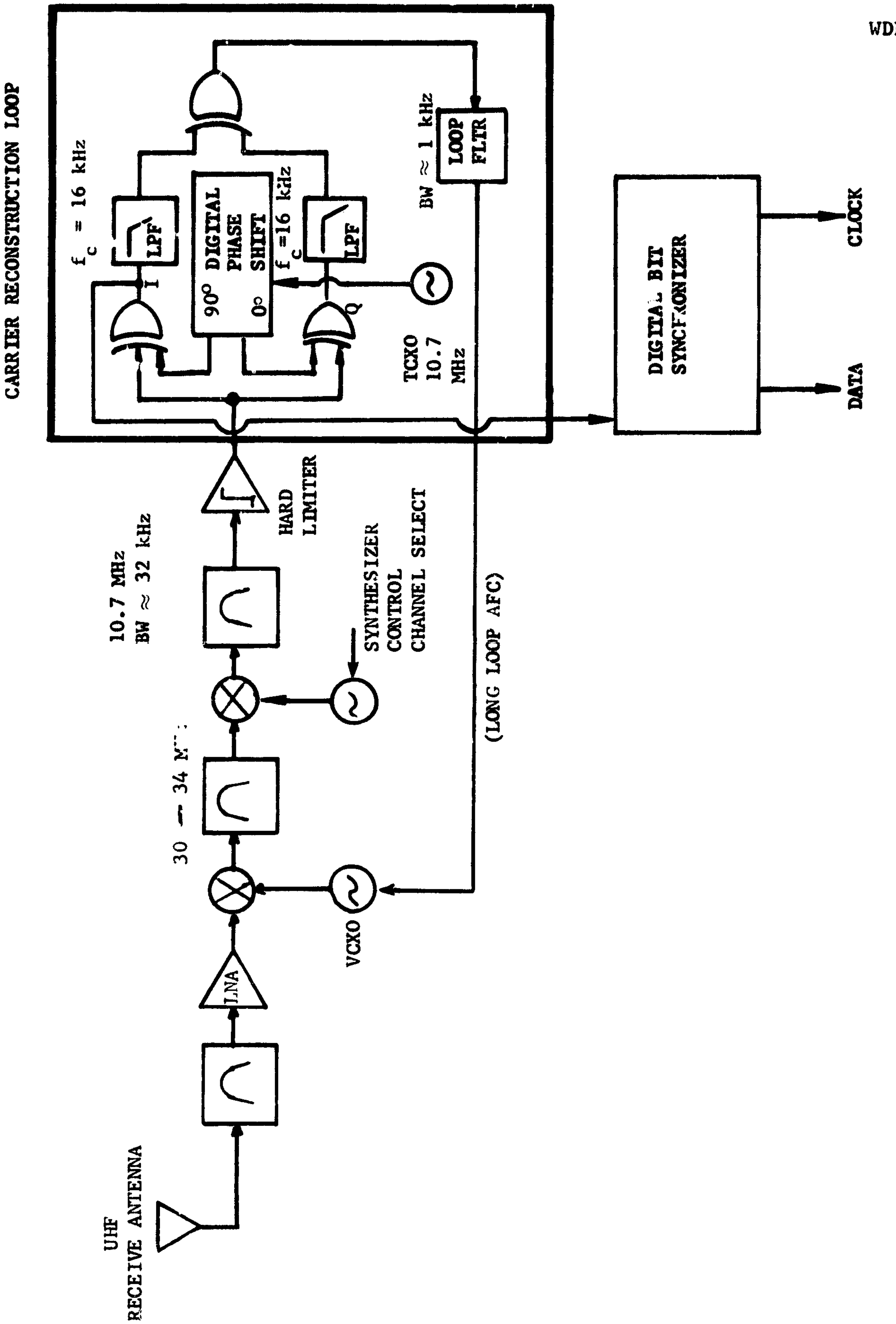


Figure 8.3-3. Receiver Data Demodulator and Bit Synchronizer

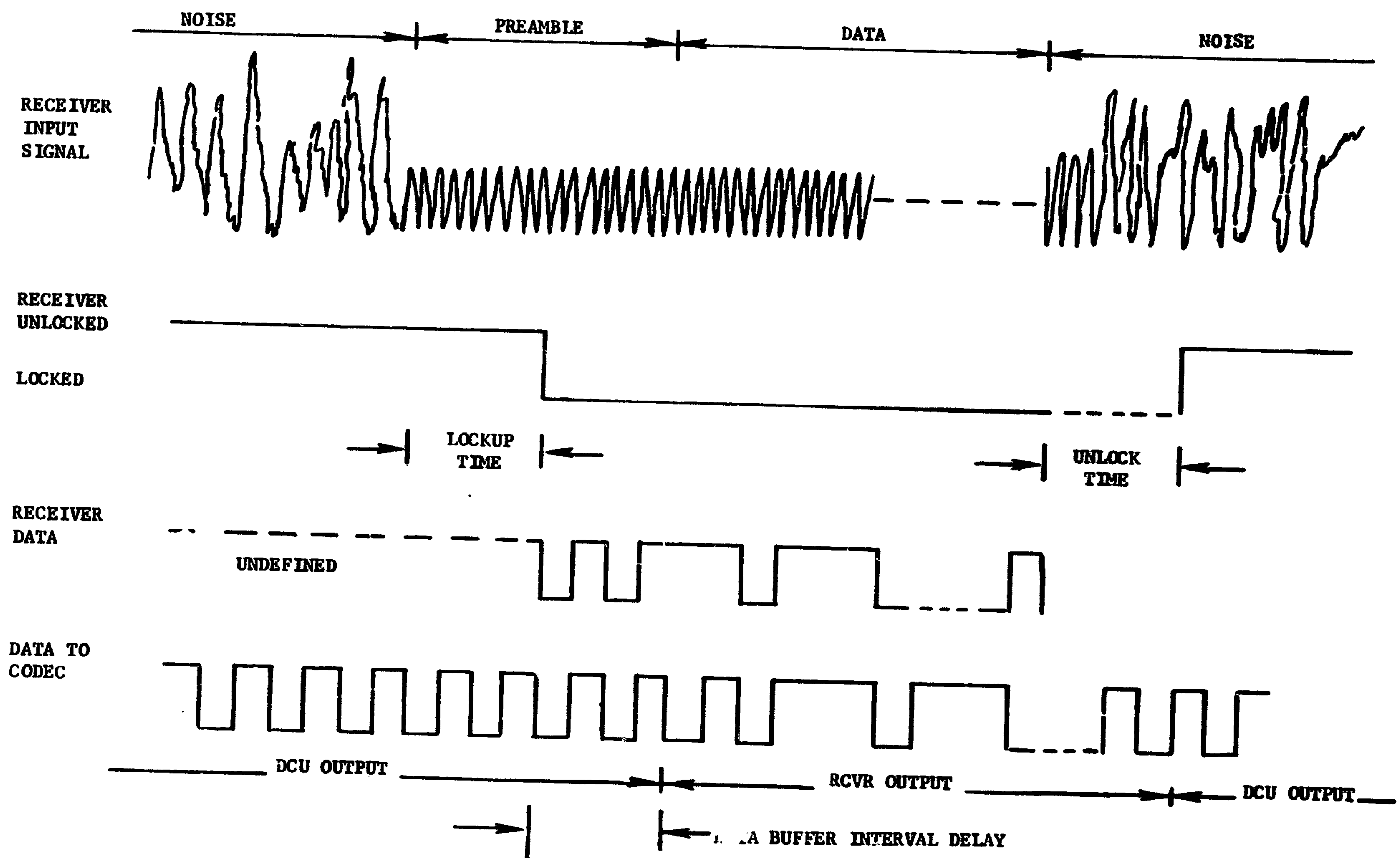


Figure 8.3-4. Receiver Timing Diagram

signal has a higher noise component. These effects combine to decrease the average phase lock loop gain, and consequently, the noise bandwidth. Thus, the PLL is somewhat adaptive to input SNR.

Actual measurements (Ref 8-1) of BER versus E_b/N_o performance of a breadboard version of this digital scheme indicate less than 1.2 dB departure from theorectical performance down to an E_b/N_o of 4 dB (data was not taken below 4 dB).

The bit synchronizer is also a digital-implemented circuit that has been breadboarded and tested. Operation of the bit synchronizer has been tested down to an E_b/N_o of 2 dB. The bit synchronizer circuit in particular lends itself to LSI implementation, which would be highly desirable for large production.

8.3.5 Receiver VOX Operation. Figure 8.3-4 illustrates the timing diagram for the receiver. Since the receiver requires a finite time to acquire an arriving carrier, the transmitted preamble must be longer than the longest expected receiver acquisition time. Using a preamble with a 1010---pattern for fast bit synch acquisition, the delta modulator codec would normally output silence during the preamble transmission interval. For this reason, before receiver lock-up occurs, the digital control unit routinely sends a constant 1010--- pattern to the receiver codec. Correspondingly, the receiver data stream is then sent to the codec after lock-up occurs.

In addition, the receiver takes a finite time interval to decide (typically 1 to 6 msec) when a received carrier has dropped out or disappeared. This decision interval is shown as the unlock time in Figure 8.3-4. Data buffering is required to prevent random data (from demodulating noise) from entering the codec and being converted to a noise burst before the receiver is squelched (DCU outputs a 1010--- pattern to the codec). The data buffer interval allows the received data to be delayed so that the receiver can be squelched before the random data is sent to the codec. A typical data buffer interval would be from 6 to 10 milliseconds.

8.3.6 Voice Data Inversion. If data inversion is to be avoided, coherent detection of BPSK signals requires resolution of the phase ambiguity that can occur in the

in the carrier reconstruction circuit. Resolution normally requires use of a known sync word or preamble at the start of the transmission. In a delta modulation system, data inversion is immaterial since the effect is to change the slope of the baseband signal from positive-going to negative-going or visa versa. Since the ear cannot tell the difference, there is no necessity to resolve the inversion if it occurs.

It should be noted that while data inversion normally lasts the entire time the carrier is present, if the carrier reconstruction circuit should lose lock momentarily (for reasons outlined in Section 8.3.3) and then relocks 180° out of phase from the previous lock point, a disturbing "glitch" will be heard in the user headset. If the carrier reconstruction circuit flips phase frequently during a transmission then conversations may become impossible.

8.3.7 Transmit/Receive Antenna. At present this study has not identified a single antenna having the required radiation patterns and bandwidth that allows operation at both 850 MHz and at L-band. Several antenna development and manufacturing sources were contacted for advice. However, for the moment the only solution appears to be separate antennas for transmit and receive.

Of those available, the best antenna type is considered to be the bent monopole, turnstile antenna described in the demonstration experiment plan. Including a ground plane such an antenna would be about 2 feet wide and 8 inches high for the 850 MHz band and about half this size for the L-band version. Thus, the two antennas required could be easily mounted on top of a mobile terminal, such as the roof of a car.

8.4 Low-Rate Data Channel

The mobile terminal can also handle low speed ($\leq$16 kbps) data as contrasted with voice data. The actual operating data rate to and from the terminal would remain at a 16 kbps rate and the DCU would perform the necessary digital processing, such as data bit averaging, to translate the 16 kbps rate to a lower rate. Thus, the BER encountered at 16 kbps would be considerably improved at the lower data rate due to the bit redundancy added to the system.

A simple hardware implementation of the low-rate data channel is shown in Figure 8.4-1. The UART's shown are standard IC's available off-the-shelf at low cost (transmitter and receiver in one IC at $7 each in unit quantities). The UART acronym stands for Universal Asynchronous Receiver Transmitter. The IC transmitter accepts any parallel input of 5,6,7, or 8 bits (programmable) and outputs a serial data stream with a bit rate of 1/16 the clock rate, automatically inserting stop and start bits. Similarly the receiver accepts an asynchronous data stream and converts it to a 5,6,7, or 8 bit (programmable) parallel output.

A typical data word in asynchronous format would consist of 10 bits per word; one start bit, seven bits of data, one parity bit, and one stop bit. For example, the transmit data from the user terminal at 2400 bps is equivalent to 240 words a second. In order to convert the transmit data stream at 2400 bps to a redundant 16 kbps data stream, a data rate of 2666.7 bps is used for the UART XMTR, a data rate exactly 1/6 of 16 kbps. Consequently, 6 bits of data at 16 kbps correspond to 1 bit of data at 2666.7 kbps.

By using a 6:1 redundance it is possible to use majority voting to correct for errors in the data stream and up to 2 errors in 6 bits can be corrected with 100% certainty. However, such a majority voting technique would require precise information about the starting point of the word, which may not be known due to a possible error on the start bit. It would be possible to force the start of every word transmission at exactly 6 bit multiples of the 16 kbps data stream so that known starting points exist for the receiver to work with. This scheme allows a bit sync circuit in conjunction with an error correction circuit to always find the starting point of every word so long as no more than 2 errors occur in every 6 bits of 16 kbps data that comprised one start bit at 2666.7 bps.

Since data is arriving from the user terminal at 240 words per second and can go to the biphase modulator at 266.67 words per second, there is no possibility of data overflow in the UART's since words can be transmitted faster than they are received. Note that word rate to the biphase modulator is actually at a 240 word per second rate even though it is capable of faster transmission rate. For this reason the receiver also cannot overflow since words are received and transmitted at a 240 word per second rate.

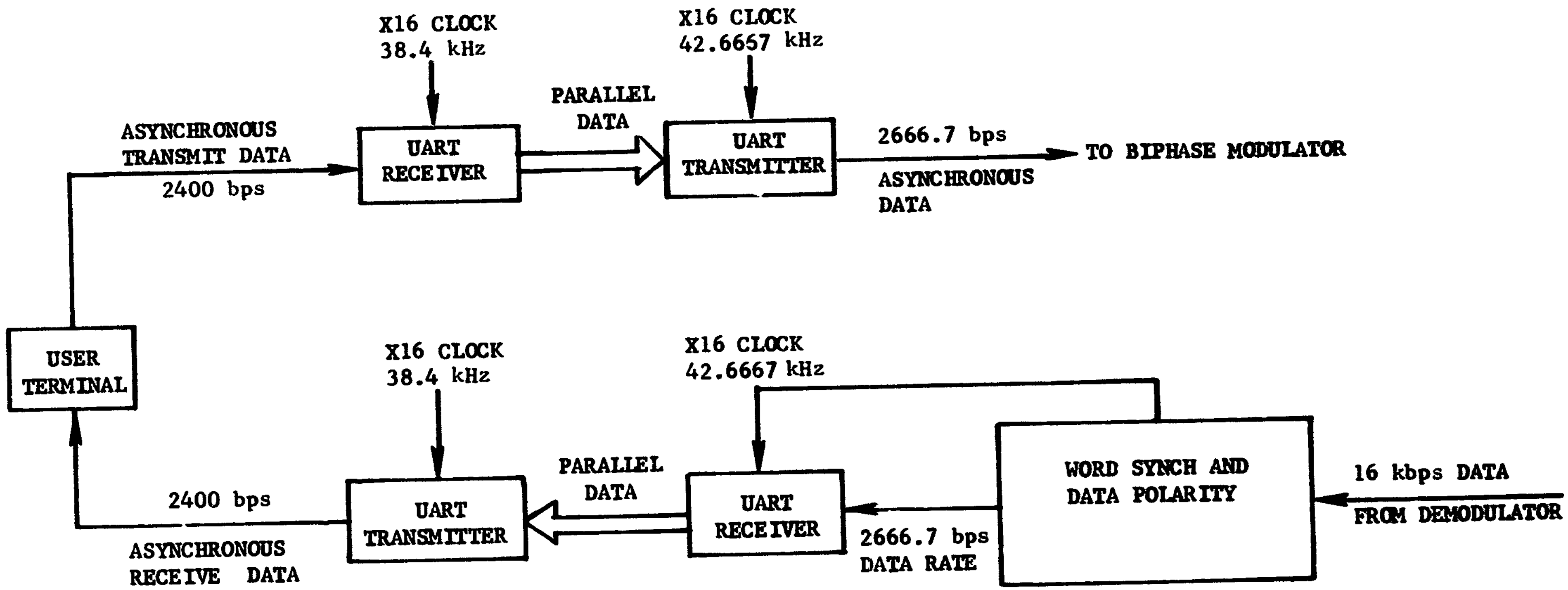


Figure 8.4-1. Low Rate Data Channel

Figure 8.4-2, Receiver Word Synch and Data Polarity Correction Circuit, is a block diagram of a circuit that will accept 16 kbps data encoded at a baseband word rate of 6 bits per word stream and generate a 2400 word per second data stream. (Note: each word is a "1" or "0" and would correspond to a bit at the lower data rate. However, "word" is used here to distinguish the lower data rate bits from the 16 kbps data bits).

Operationally the circuit makes a majority vote decision on the six bits comprising each word and then outputs a "1" or "0" decision to be transmitted as the lower data rate bit. Word synchronization is required to ensure that only the correct 6 bits comprising the transmitted word are used for a voting decision and is accomplished by looking for a 000000111111 received 16 kbps data pattern and generating a sync pulse when it occurs. This particular pattern occurs whenever a lower data rate 0/1 word pattern is transmitted. If any 0/1 word pattern is received with errors then it would not generate a synch pulse. However, if a 0/1/1 pattern were transmitted and received as shown in Figure 8.4-3, then a sync pulse would

TRANSMITTED WORDS	0 | 1 | 1
TRANSMITTED BITS	000000111111111111
RECEIVED BITS	000000011111111111
CORRECT SYNC PULSE	1
GENERATED SYNC PULSE	1

Figure 8.4-3. Word Sync Error Generation

be generated that was delayed one clock time. This would degrade the error correction capability until a 0/1 pattern occurred again and generated a correct synch pulse. Circuitry could be added to prevent this problem but in the interest of simplicity is omitted.

Since the receiver demodulator does not resolve data inversion plus the fact that degraded terminal operation may cause occasional inversion of data, the

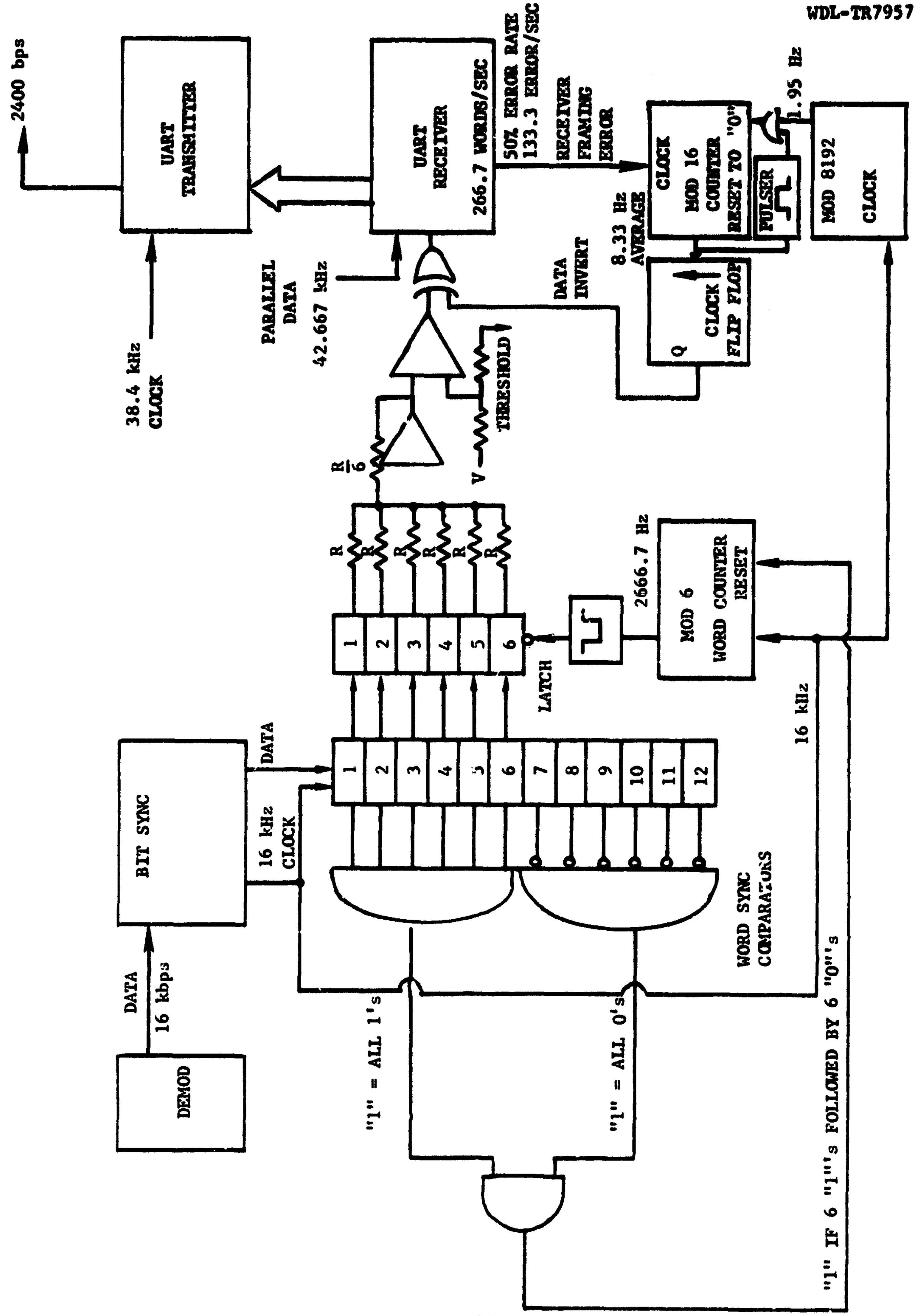


Figure 8.4-2. Receiver Word Sync and Data Polarity Correction Circuit

data should be transmitted at 16 kbps asynchronously to and from the terminal. Asynchronous transmission will always have start and stop bits that would be used to resolve data inversion. The UART would be used to detect data inversion by the framing error signal it generates when a proper stop bit is not received. Figure 8.4-2 shows a data polarity correction circuit consisting of a UART receiver, MOD 16 counter, a flip-flop, and exclusive OR gates. If data is inverted then the UART receiver would be receiving 266.7 words a second, and would generate a framing error on half of the words, assuming random distribution. Thus, 133.3 framing errors per second would be generated. The MOD 16 counter would generate a pulse after 16 framing errors that would trigger the flip-flop to change state, which causes the exclusive OR gate to invert the data. Simultaneously, the MOD 16 counter would clear itself. After the data has been corrected to the right polarity the UART should frame correctly within 10 words, thus preventing the flip-flop from being pulsed by the MOD 16 counter and inverting the data again. The MOD 16 counter would be cleared every 8192 clock pulses (or about every 1/2 seconds) to prevent random framing errors from being indefinately accumulated.

8.5 DAMA Network

Ultimate mobile system flexibility requires the capability for any station in the network to be able to easily access any other station in the network. For a small number of stations in the network a manual DAMA function could be accomplished by requiring each terminal to have a preassigned receive frequency. However, a full automatic DAMA system would be required to accommodate a large number of remote stations; that is, whenever the number of stations exceeds the number of available channel frequencies assigned to each spot beam. Table 8.5-1 lists the pertinent DAMA system requirements. One of the key items listed is the operational requirement that the DAMA system be transparent to the user. In point of fact, the DAMA system should process calls in a manner that the user is familiar with; for example, the commercial telephone system. Section 8.1.2 briefly outlined the operational scenerio proposed for the DAMA operation, which to the user appears as a conventional telephone network. A more detailed DAMA call sequence is described below in the next section.

Table 8.5-1. DAMA System Requirements

TRAFFIC MODEL

- SERVES 10,000 STATIONS
- TOTAL AVERAGE CALL ORIGINATIONS - 180 PER SECOND
- AVERAGE CALL DURATION - 11 SEC/CALL

OPERATIONAL REQUIREMENT

- TRANSPARENT TO USERS

DESIRABLE CHARACTERISTICS

- MINIMIZE POST DIALING DELAY
- MINIMIZE NUMBER OF SIGNALING CHANNELS
- MINIMIZE OVERALL SYSTEM COST
- CAPABLE OF SYSTEM CLEARING WHEN OVERLOADED
- CAPABLE OF PRIORITY ASSIGNMENT AND BREAK-IN

The DAMA configuration, as highlighted in Table 8.5-2, is a centralized system using a single earth station for network control. The network control station is referred to as a Master Control Station (MCS). The MCS uses a mini-computer system to control all stations, assign frequencies and maintain status of all terminals and channels in every spot beam. To minimize the total network cost, each remote station is provided with only one modem which is used for both signalling with the MCS and for communicating with other mobile terminals.

Table 8.5-2. DAMA System Configuration

- DAMA NETWORK - CENTRALIZED
- CALL REQUEST - RANDOM ACCESS/UNSLOTTED
- USER TERMINAL - SHARED MODEM FOR CHANNEL REQUEST/ ASSIGNMENT AND COMMUNICATION (16 kbps)

Communications between the user terminals and the MCS is via data channels to/from the MCS and each terminal as shown in Figure 8.5-1. The MCS broadcasts to the terminals on a Broadcast Channel (BCC) dedicated to the MCS. The user terminals transmit assignment request to the MCS on a Common Signalling Channel (CSC) shared by a number of terminals. Random access is used for the user terminal-to-MCS communications link. Any terminal desiring a channel assignment transmits a service request data block to the MCS. If the MCS receives the data block (format shown in Figure 8.5-2) without error, the MCS acknowledges back to the terminal using the MCS broadcast format. In the event the MCS does not receive an error-free block, it ignores the message and does not acknowledge. If the user terminal does not receive an error-free acknowledgement within a fixed time interval, it retransmits the service request to the MCS after waiting a random time interval. This random time interval is necessary to prevent two or more stations from continually trying to access the MCS simultaneously. When a call is completed, the two terminals report to the MCS using the release report format of Figure 8.5-2.

The MCS terminal should have a G/T, figure-of-merit, 5 to 10 dB higher than that of the user terminals. A high G/T for the MCS ensures a very low BER on the common signalling channels from the user terminals. Since the MCS is a fixed station, a high gain antenna that requires pointing should not be a problem. The broadcast channel from the MCS to the user terminals should also be operated at a higher EIRP than the communication channels in order to maintain low BER to the user terminal. How much higher the EIRP should be depends upon the satellite transponder capacity in the regional transponders.

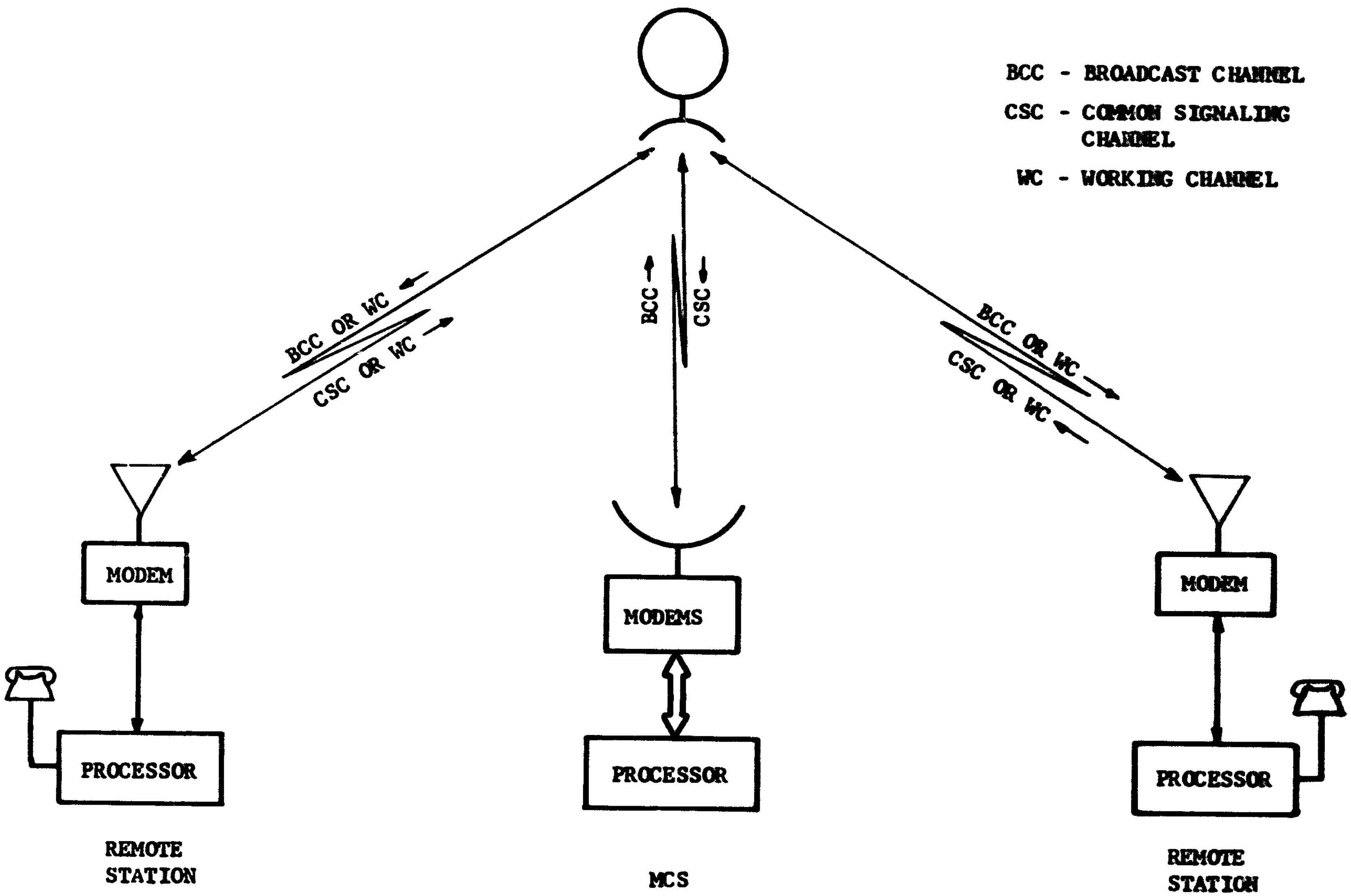


Figure 8.5-1. DAMA Network Configuration

- SERVICE REQUEST FORMAT

PREAMBLE	START OF MESSAGE	CALLING STATION ID	CALLED STATION ID	TYPE PRIORITY	CHECK BITS
8 BITS	8 BITS	16 BITS	16 BITS	8 BITS	16 BITS

- RELEASE REPORT FORMAT

PREAMBLE	START OF MESSAGE	REPORTING STATION ID	CALLED STATION ID	TYPE PRIORITY	CHECK BITS
8 BITS	8 BITS	16 BITS	16 BITS	8 BITS	16 BITS

- MCS BROADCAST MESSAGE FORMAT

START OF MESSAGE	REMOTE STATION ID	TYPE OF MESSAGE	TRANSMIT CHANNEL ASSIGNMENT	RECEIVE CHANNEL ASSIGNMENT	CHECK BITS
8 BITS	16 BITS	3 BITS	16 BITS	16 BITS	16 BITS

Figure 8.5-2. DAMA Message Format

In order to add DAMA to the mobile network a single master control station is required plus microprocessor hardware and fully synthesized modems at the user terminals. The MCS should have fully redundant equipment throughout to effectively eliminate outage time. Obviously, if the MCS malfunctions the entire network is inoperative.

The overall cost of the DAMA function would consist of development costs of 1 to 2 million dollars, plus the cost of the MCS, and the differential costs added to each user terminal for the DAMA hardware. Obviously, to be economically viable the application of DAMA requires a large user population over which to spread the costs.

8.5.1 DAMA Call Sequence. This section presents a detailed step-by-step description of a typical DAMA call sequence. The numbered steps below refer to the encircled numbers appearing earlier in Figure 8.1-4. This description also refers to the data formats for the service request, the MCS broadcast message, and the release report as shown in Figure 8.5-2. A typical operational DAMA call sequence proceeds as follows:

1. Originating user picks up the phone. The phone sends an off-hook signal to the processor.
2. The originating remote station receives and detects the off-hook condition. If the processor is ready to accept the dialing, it sends an audible dial tone back to the originating user Otherwise, it sends a busy tone back.
3. Upon hearing the dial tone, the originating user can now proceed to dial the called station number, select priority and/or type of connection (simplex or duplex, voice or TTY).
4. The originating remote station processor receives and stores the dialing information, formats a request message and transmits this message burst to the MCS using the preassigned common signalling channel. This burst transmission will be repeated if a reply is not received from the MCS after appropriate time delay. To avoid overloading the CSC, a request will only be transmitted three or four times. After three or four request transmissions with no reply, the processor sends a busy tone to the originating user.

5. MCS receives and decodes the request from the originating remote station. If both satellite channels and the remote destination station are available, the MCS selects the working channel frequencies and assigns these frequencies to the originating and the destination stations using the broadcast channel. If satellite channel and/or destination remote station are unavailable, either a busy message or a queue message can be sent to the originating remote station.
6. The originating remote station, upon receiving the WC assignment, tunes its modem to the assigned frequencies.
7. The destination remote station, upon receiving the WC assignment, tunes its modem to the assigned frequencies, rings the called user's telephone.
8. The destination station sends a ringback signal to the originating user via the working channel.
9. The called user picks up the telephone and communication may now proceed.
10. The destination remote station processor detects the off-hook condition and sends an answer signal back to the originating remote station. This signal may be used to reset a timeout clock if desired.

At the end of the conversation, the release sequence is as follows, assuming the originating user releases first. The release sequence would be identical if called user releases first with originating and destination remote station functions interchanged.

11. Originating user hangs up the phone. The phone sends an on-hook signal to the processor.
12. The processor detects the on-hook condition and sends a release signal to the destination remote station via the working channel.
13. The originating remote station processor retunes the modem to the preassigned CSC frequency, formats and transmits a Channel Release message to the MCS. This transmission will be repeated until an acknowledgement is received from the MCS.
14. MCS receives the release report from the originating remote station, sends an acknowledgement back to the originating remote station and waits for the destination remote station to report.

15. The originating remote station receives the acknowledgment and places the station in the idle (available) condition.
16. The destination remote station receives the release from the originating remote station and tunes its modem to the preassigned CSC/BCC frequencies.
17. The called user hangs up the phone and sends an on-hook signal to the processor.
18. The destination remote station processor, receiving the on-hook signal from the user telephone or after a timeout period from the time the originating user release signal is received, formats and transmits a release report message to the MCS.
19. The MCS receives the report from the destination remote station, places the working channel frequencies back into the available channel pool, and transmits a report acknowledgement to the destination station.*
20. The destination remote station receives the acknowledgement and places itself back in idle condition.

Based on the call sequence described above, the possible message formats are shown in Figure 8.5-2. A service request from the originating remote station to the MCS is a coded message 72 bits long. At a 16 kbps rate, the message duration is 4.5 msec. The format contains special bits for a preamble, a start-of-message, calling and called station ID, priority/type, and check bits. To facilitate the detection by the MCS processor, the same message length may be used to report channel release. Instead of calling Station ID, these 16 bits are used to identify the reporting station. A unique 8 bit work is used in the "TYPE" slot to identify the message to be either a request or a release message.

A master station broadcast message format, shown in Figure 8.5-2, is 80 bits long. At a 16 kbps rate, the message duration is 5 msec. The same message format, utilizing the 8-bit "type of message" slot, may be used to ask for status report, to prohibit or grant permission to transmit over the CSC, and to send other commands to the remote stations. The status report function may be used to update and correct the master station satellite channel usage/remote station status table during non-busy hours or to re-establish the table in case of MCS failure. Prohibition of transmission over the CSC may be used to clear the CSC channels in case of overload.

*The type of "smart" DAMA system described here also provides for operational malfunctions, such as an operator fails to hang up or a terminal fails, by a combination of MCS microprocessor safeguards and MCS operator alerts and override functions.

8.6 Cost Estimate

The cost model for the mobile terminals covers cost estimates for quantities of 10, 100, 1000 and 10,000 terminals. However, the detailed terminal hardware configuration differs significantly at each quantity level due to the necessity to minimize the total of non-recurring and recurring costs. For example, to exploit LSI and hybrid technologies requires a large initial investment. Since the initial investment would be spread over the total number of terminals produced, LSI and hybrid circuitry could economically be utilized only for large production runs where the per unit cost is low. Similarly, the same is true of DAMA since a large investment is necessary for provision for the Master Control Station. A cost estimate summary is presented in Table 8.6-1 showing the cost per terminal for voice only and the incremental costs of adding low-speed data and DAMA.

Table 8.6-1. Cost Estimate Summary
(per mobile terminal)

NUMBER OF UNITS:	10	100	1000	10000
VOICE ONLY	$64,380	$20,946	$5,756	$3,947
ADD'L FOR DATA	$ 2,600	$ 440	$ 230	$ 163
ADD'L FOR DAMA	$250,500	$25,500	$3,070	$ 447
DATA + DAMA	$253,100	$25,940	$3,300	$ 610
VOICE/DATA/DAMA	$317,480	$46,886	$9,056	$4,557

8.6.1 Terminal Population of 10. Low production runs of 10 or less terminals would use off-the-shelf hardware wherever possible. However, most conventional equipments available are not designed to work in the environment associated with mobile operation; that is, small physical size, D.C. operation, VOX operation at E_b/N_o values as low as 3 dB, and high shock and vibration. The modem itself presents the greatest concern and probably would require a total non-recurring cost of $300,000 to develop a modem more in line with the mobile terminal needs. This high cost is difficult to justify for only 10 terminals. DAMA operation is judged to be economically out of the question for populations of only 10 terminals.

Table 8.6-2 details the overall cost breakdown expected for a production run of 10 units. The costs are developed from assuming that only 10 units are produced and then the production line is shut down. Obviously, high setup, shut down, and design costs make such small production runs very uneconomical (amortizing recurring and non-recurring costs over 10 units makes a per unit cost of $64,380). Note that the costs developed do not include any profit, administrative overhead, and distribution costs that would be added by a manufacturer, and the final sell price would probably be 20% higher than the stated costs. A separate cost breakdown estimate for the modem alone is presented in Table 8.6-3.

8.6.2 Terminal Population of 100. A production run of 100 terminals is large enough to start achieving some economy of scale in production. The terminal design would be essentially the same as the design for a production run of 10 since the quantity is still too low to justify significant design changes. However, the primary per unit savings occur as a result of higher volume buys on parts and more units over which to amortize the non-recurring costs. In general, a 30% savings can be expected on a production run of 100 units. Thus the expected costs per terminal would be $16,926 recurring and $4,020 non-recurring for a total of $20,946 per terminal.

8.6.3 Terminal Population of 1000. Production runs of 1000 mobile terminals or greater would start to justify the use of LSI and hybrid technologies. Table 8.6-4 shows the recurring and non-recurring costs expected for a production run of 1000 terminals. The costs are strictly estimates only since component manufacturers were found to be reluctant to quote large quantity prices without a firm possibility of an order (most items required are not in large volume production such as HPA's, LNA's, etc.; plus most circuitry would be in hybrid form rather than in discrete components). Furthermore, the terminal design is not known in sufficient detail to specify circuit requirements at this time.

For this production level the non-recurring costs are generally higher than the low-quantity costs (of Table 8.6-1) in order to achieve designs capable of volume production. The resulting per unit terminal costs for 1000 units is

$4,660 recurring and $1,096 non-recurring, for a total of $5,756. The DAMA non-recurring costs are identical to the low production costs since a single Master Control Station (MCS) is required in all cases and the software/hardware is essentially the same. DAMA recurring costs for the MCS assumes a fully redundant station in all cases but the larger user populations require additional common signalling channel transmitters, receivers, computer I/O ports, and larger transmit amplifiers to handle more carriers.

8.6.4 Terminal Population of 10,000. A production run of 10,000 mobile terminals should achieve a per unit recurring cost reduction of 20 percent over the per unit recurring costs for 1000 units based on 10% volume buy savings and 10% product redesign. The non-recurring costs are estimated to double for product redesign savings. Thus the per unit terminal costs for 10,000 units would be $3,728 recurring and $219 non-recurring, for a total of $3,947. If all 10,000 terminals also included DAMA and low-speed data capability, the added costs are expected to be $610 per terminal, for a net terminal cost of $4,557.

Table 8.6-2. Cost Estimate: Production Quantity of 10

	PER UNIT RECURRING	TOTAL NON-RECURRING
Transmit Antenna	$1,000	$6,000
Receive Antenna	$1,000	
HPA (40W at L-Band, Saturated)	$3,350	
Receive Filter	$1,100	
Transmit Filter	$1,200	
Transmit Isolators	$ 130	
Receive Isolators	$ 130	
Receive Mixer/Amp	$ 900	
Receive Local Oscillator	$ 550	
Transmit Transponder Synthesizer	$1,000	$40,000
DCU	$ 200	$18,000
Delta Modulator Codec	$ 75	
HPA Power Supply	$ 500	
Modem Power Supply	$ 150	
Power Term for HPA	$ 100	
Receiver IF Filter	$ 200	
Miscellaneous (PAD's, cables, etc.)	$ 250	
Modem	$6,345	$300,000
Terminal Integration & Checkout	$6,000	
Terminal Design & Production Startup & Shut down Costs		$36,000
Test Plan		$2,000
TOTALS	$24,180/unit	$402,000
Options:		
Low Speed Data (RS232 Interface)	$ 200	$24,000
DAMA		
Terminal Unit Cost	$ 500	$2,000,000
Master Control Station	$500,000	

Table 8.6-3. Modem Cost Breakdown Estimate

MODEM	PER UNIT RECURRING
SYNTHESIZER RX	$ 75
FILTER	$ 50
RX IF	$ 50
DEMOD	250
BIT SYNC	75
SYNTHESIZER TX	75
BIPHASE MOD	20
CHASSIS	500
POWER SUPPLY	100
REF OSC	100
CXR CTRL	50
	$1,345
ASSY 200 HRS @ $20/HR	$4,000
TEST TIME 40 HRS @ $25/HR	$1,000
TOTAL	$6,345

Table 8.6-4. Cost Estimate: Production Quantity of 1000

	PER UNIT RECURRING	TOTAL NON-RECURRING
Transmit Antenna	$ 150	$ 10,000
Receive Antenna	150	10,000
HPA/Filters/Isolators	200	10,000
Receive LNA/Mixer/Amplifier	300	24,000
Receive Local Oscillator	200	
Transmit Transponder Synthesizer	300	54,000
Transmit Mixer/Filter/HPA/Filter	400	40,000
Terminal Power Supply	100	
DCU	40	18,000
Receive Filter	80	
Modem	2,400	750,000
Miscellaneous	100	
Terminal Integration & Checkout	240	100,000
Terminal Design & Production Costs		80,000
TOTAL	$4,660	$1,096,000
Options		
Low Speed Data	$ 200	$ 70,000 (DAMA software changes) $ 30,000 (Hardware design)
DAMA		
Terminal Unit Costs	$ 200	$2,000,000
Master Control Station	$800,000	

9.0 FIELD DEMONSTRATION EXPERIMENT PLAN AND COST ESTIMATE

The objective of this task involves planning a field experiment and demonstration to verify the feasibility of a digital mobile voice communication satellite system using the voice digitization and digital modulation techniques selected by the previous considerations and investigations. In addition to the technical planning aspects of the demonstration, a costing exercise was undertaken to estimate the cost of carrying out a field experiment to measure performance for a range of system parameter values. Narrowband FM modulation is included in order to compare the digital approach to the conventional analog FM-SCPC scheme.

A baseline taken for this experiment plan calls for the use of off-the-shelf conventional rack-mounted mobile equipment wherever possible to minimize costs. This baseline was followed throughout to the point where only two items, the mobile antenna and the digital baseband modem, require minor development work. However, this development consists mainly of retrofitting or modifying existing hardware to meet the specifications of the experiment. Otherwise, all of the experimental equipment is standard, utilizing present-day solid-state technology.

For the field demonstration experiment the NASA ATS-6 experimental satellite is planned to be utilized using the L-band pencil beam satellite antenna. The experiment involves two mobile earth terminals, vehicularly mounted, and primarily designed to verify and evaluate a voice-operated, digital mobile communication system.

Completion of the full field experiment is expected to require 8 to 11 months, depending upon the scope of the actual field testing.

The total cost of carrying out the experiment plan described here is estimated to be about $170,000 including hardware and labor. However, under an option described in the detailed cost breakdown, an additional cost of $14,000 could be added to this figure if the option were exercised. No cost has been assigned for the use of the satellite other than a small cost for satellite scheduling consisting mainly of an administrative function.

9.1 Experiment Plan/Test Parameters

The field demonstration experiment plan is divided into three distinct phases consisting of:

- Phase 1 - The system design, procurement, and hardware implementation of two mobile communication earth stations operating at L-band.
- Phase 2 - In-lab checkout and pre-testing of the earth stations to establish a performance reference as a preparation for field testing.
- Phase 3 - The actual field testing, experimentation, and evaluation of the data obtained from the field tests.

For the field demonstration tests, the ATS-6 satellite would be utilized in the 1535 to 1565 MHz receive band and in the 1630 to 1670 MHz transmit band, making use of the L-band pencil beam satellite antenna. During the field testing, several system parameters and quality factors would be evaluated including:

- Link margins, implementation margins, and antenna performance
- Bit error rate, E_b/N_o, and performance degradation at low carrier to noise density ratios
- Earth station system Figure of Merit, G/T, and carrier to noise density ratio, C/kT
- Informal speech intelligibility assessment and comparative reception quality rating for both the digital and the analog FM methods

The results of the field test data would be reviewed, evaluated, and compiled into a final report describing the equipment, test procedures, and recommendations.

Figure 9.1-1 illustrates the experimental hardware configuration in block diagram form. In the experimental setup the receive-chain and the transmit-chain are coupled into a common antenna via a diplexer. The antenna is mounted on top of a vehicle above a ground plane with the RF and baseband equipment installed inside the vehicle. The baseband equipment is designed to operate at 70 MHz which is considered a standard interface frequency. Aside from the field equipment, one test translator is required for loop testing and checkout of the digital radio units in the lab prior to experimental testing in the field.

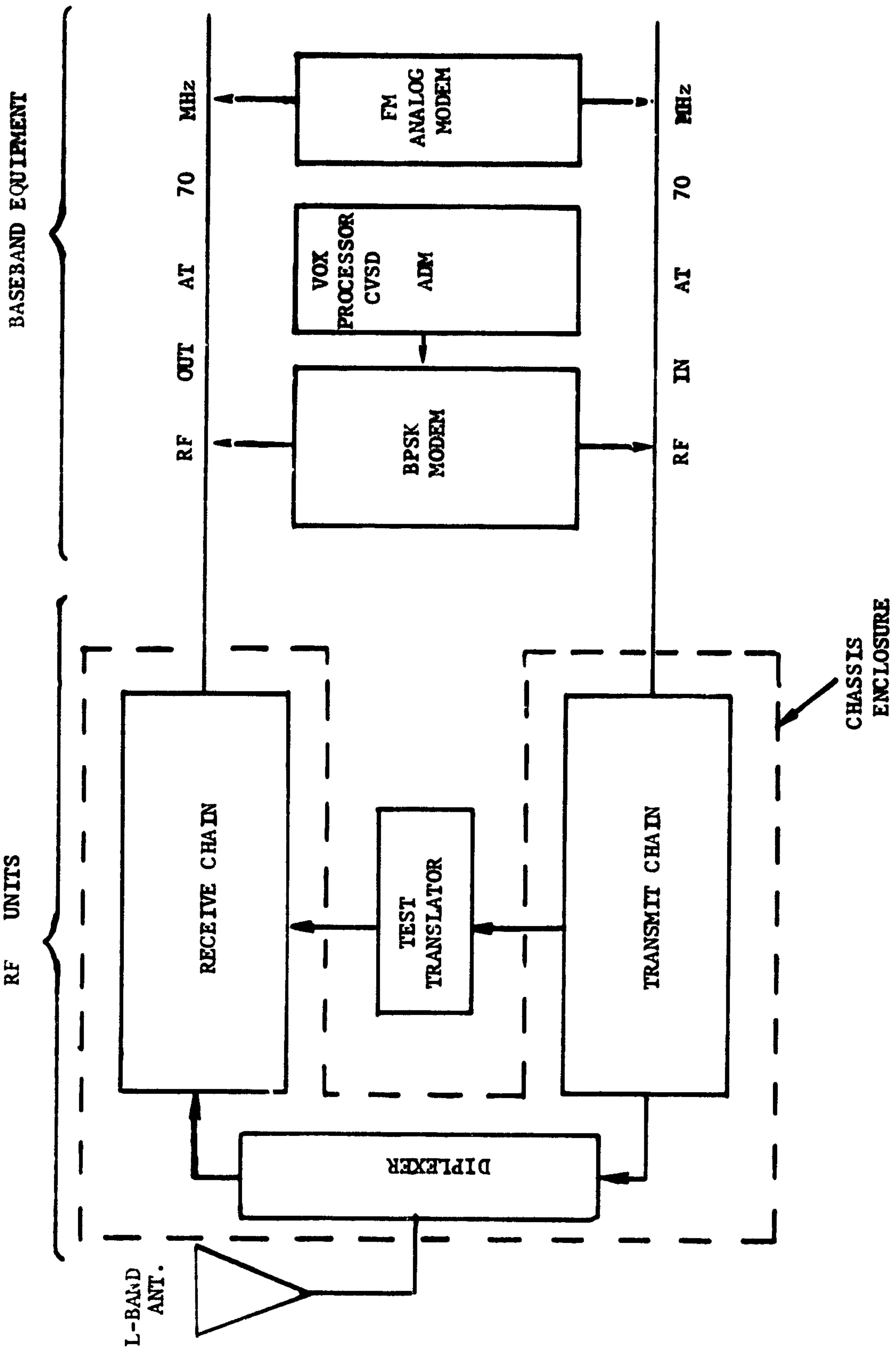


Figure 9.1-1. Field Experimental Hardware Configuration

9.2 Hardware Requiring Development

In attempting to conform closely with the ground rules and objectives of this experiment, a thorough search was performed for potential vendors from which to procure off-the-shelf or near off-the-shelf conventional equipment, as much as possible. In fact, the resulting equipment configuration was designed around these available components for system parameters such as gains, noise figure, power levels, etc. However, two items were determined to require some degree of development. These are the antenna for the RF equipment and the modem in the baseband equipment.

9.2.1 L-Band Antenna. Only one out of ten antenna companies contacted was found that manufactured an antenna which is close to meeting the requirements of this experiment. Even this antenna requires some modification; however, the cost of modifying the existing design is relatively small and the end-result is an omni-directional, hemispheric antenna suitable for volume production. This particular antenna is the direct result of many years' work with hundreds of thousands of dollars already spent in the development. The additional cost of retrofitting the antenna for the demonstration experiment is considered a small price to pay for superior performance. A detailed technical description of the antenna is presented in Section 9.4.2.

9.2.2 PSK Modem. In spite of thorough canvassing of existing vendor's available products of this type, no suitable off-the-shelf or near off-the-shelf PSK modem was found to be available from outside vendors for mobile terminal speech operation using VOX. The main reason for this lack of suitable modems is the stringent requirement of quick acquisition and reacquisition time, 2 to 4 msec, at low E_b/N_o values (3 dB). Most of the vendor's available modems are designed to operate at high E_b/N_o values of around 8 dB minimum and require about 20 to 30 seconds to lock-up. A possible solution to this dilemma is to use an existing SCPC modem that was developed by Ford Aerospace and Communications Corporation (FACC) and utilized in other deliverable equipment* successfully in the past. This modem would required certain minor modifications for the intended application. For voice digitization it would be necessary to either purchase or build the voice digitization equipment

*Indonesian DOMSAT Program

so as to directly interface with the modem. Since a suitable alternate outside source is not available at this time, the costing estimate will be based upon use of this existing SCPC modem.

9.3 Cost Summary

A complete cost baseline and a detailed cost breakdown are described in Section 9.7. The final cost estimates including hardware and labor are summarized as follows:

Phase 1 - Design and hardware implementation of two mobile earth stations		\$118,100
Phase 2 - Laboratory checkout and verification tests		\$ 12,600
Phase 3 - Field verification experiments		\$ 38,800
Miscellaneous		\$ 500
	TOTAL	\$170,000

These cost estimates are based on two complete baseband modems loaned to NASA free-of-charge for 3 to 4 months, which is the estimated length of time to complete Phases 2 and 3 of the experiment. If NASA exercised the option to purchase the modems, the additional cost would be about \$14,000 for a total of \$184,000.

The total estimated time to complete the field demonstration experiment objectives is between 8 and 11 months.

9.4 Technical Description of RF Unit

The proposed experimental mobile terminal described herein has the following special features:

- Functions with full duplex BPSK Modem
- Utilizes omnidirectional antenna with 1.8 dB minimum gain and 160^{o} hemispheric coverage, making it insensitive to terrain irregularities

- Ample filtering employed to reduce problems due to narrow separation of transmit and receive frequencies
- Capability to operate at any frequency within the ATS-6 satellite transmit L-band (1535-1536 MHz) and receive L-band (1630-1670 MHz) by utilizing a tunable LO and wideband bandpass filters
- AFC provided in the LO for convenience of fine tuning and frequency locking
- Directional couplers provided at the transmit output and receive input ports to facilitate RF loop testing
- Test translator provided as part of the RF equipment for RF loop testing in the laboratory or in the field
- Transmit and receive power levels are adjustable
- Ample margins provided to assure good performance under severe external conditions

All the above features contribute to a RF unit that can perform quite well under variable conditions of weather, terrain, and localities. Table 9.4-1 presents a comprehensive listing of the detailed technical specifications and key parameters of the mobile digital radio system. Supportive derivations of system G/T and effective radiated noise power are contained in Appendices A and B.

9.4.1 Link Analysis and System Margins. In carrying out the computation of system margins, many of the key uncertainties existing in the experimental system were taken into account, particularly those that result in significant degradation of performance. The link analysis calculations are based on the following model:

- ATS-6 L-band pencil beam antenna utilized (See Table 9.4-2)
- Line-of-sight path loss used as a worst case condition
- Two test carriers present in the satellite downlink, sharing the total satellite EIRP
- Antenna noise temperature assumed at 250°K, a very pessimistic estimate even for worst case. (Note: true antenna noise temperature TBD during the measurement of G/T)

Table 9.4-1. Mobile Digital Radio Specifications

EFFECTIVE RADIATED SIGNAL POWER	16 dBW
TRANSMIT FREQUENCY (OUTPUT)	SELECTABLE BETWEEN 1630 and 1670 MHz
RECEIVE FREQUENCY (INPUT)	SELECTABLE BETWEEN 1535 and 1565 MHz
SYSTEM G/T	-27 dB/^{o}K
TYPE OF CONVERSION	SINGLE CONVERSION BETWEEN L-BAND & 70 MHz
TYPE OF ANTENNA	BENT MONOPOLE TURNSTILE HEMISPHERIC
FIELD OF VIEW (FOV)	10° TO 170° IN ELEVATION
FOV GAIN	1.8 dB MINIMUM
ANTENNA INPUT TO RECEIVER OUTPUT GAIN AT 70 MHz (EXCL. BASEBAND UNIT)	81 dB NOMINAL
TRANSMITTER INPUT TO ANTENNA OUTPUT GAIN AT 70 MHz	71 dB NOMINAL
RECEIVE OUTPUT AT MODEM INTERFACE	
• FREQUENCY	70 ± 15 MHz
• POWER LEVEL	-35 dBm +5, -15 dB ADJ.
• IMAGE REJECTION	90 dB MIN
• TRANSMIT LEAKAGE	-70 dBm
• IN-BAND SPURIOUS (DISCRETE)	-50 dBm MAXIMUM
• OUT-OF-BAND SPURIOUS	-60 dBc MAXIMUM
• C/kT	54.2 dB-Hz MINIMUM (TWO CARRIER OPERATION)
• IMPEDANCE/VSWR	50 OHMS, 1.3 MAXIMUM
• GAIN VARIATIONS OVER 1535-1565 MHz (EXCL. ANTENNA)	2 dB p-p MAXIMUM
BASEBAND PERFORMANCE	
• TYPE OF OPERATION/DATA RATE	BPSK/16-kbps, VOICE/TTY
• CARRIER ACQUISITION TIME	4 msec MAXIMUM
• DESIGN POINT BER	10^{-3}

Table 9.4-1. Mobile Digital Radio Specifications (Continued)

• THRESHOLD BER	10^{-2}
• USABLE END POINT	TBD (During Field Testing)
• DESIGN POINT LINK MARGIN	2.9 dB
TRANSMIT OUTPUT CHARACTERISTICS AT DIPLEXER INPUT	
• OUTPUT POWER	15.7 dBW, ADJUSTABLE + ZERO TO -5 dB
• 1 dB BANDWIDTH	1630-1670 MHz
• MAXIMUM OUT-OF-BAND SPURIOUS LEVEL	-16 dBm
• MAXIMUM IN-BAND SPURIOUS LEVEL	-60 dBc
• GAIN FLATNESS, 1630-1670 MHz	0.5 dB WINDOW
DIPLEXER CHARACTERISTICS	
• TRANSMIT TO RECEIVE ISOLATION	20 dB
• TRANSMIT TO ANTENNA PORT INS. LOSS	0.3 dB
• ANTENNA TO RECEIVE PORT INS. LOSS	0.4 dB
• VSWR, ALL PORTS	1.25 MAX
• IMPEDANCE	50 OHMS NOMINAL
LOCAL OSCILLATOR SPECIFICATIONS	
• FREQUENCY	
TRANSMIT CHAIN	1700-1770 MHz
RECEIVE CHAIN	1465-1495 MHz
• OUTPUT POWER	24 dBm $\pm$ 1 dB
• FREQUENCY STABILITY	$\pm$ 5 ppm. CAN BE STABILIZED BY FREQUENCY OR PHASE LOCK
• SPURIOUS SIGNALS	
IN-BAND	-100 dBc
OUT-OF-BAND	-30 dBc
• DC/INPUT POWER	20 V at 300 MA
• AFC RANGE (FINE TUNING)	$\pm$ 0.15% FREQUENCY TUNING FOR 4 TO 16 V TUNING VOLTAGE

Table 9.4-2. ATS-6 Satellite L-Band Pencil Beam Mode Parameters (Typical)

G/T	4.3 dB/^{o}K
EIRP	50.1 dBW
Antenna Field of View	$1^{o} \times 7.5^{o}$ off axis
Receive Frequency	1650 $\pm$ 20 MHz
Transmit Frequency	1550 $\pm$ 15 MHz

Typical ATS-6 satellite parameters are shown in Table 9.4-2. These parameters were used to compute the demonstration experiment link parameters listed in Table 9.4-3. Included in these calculations were all of the contributing uncertainities in both the up-link and the down-link, resulting in a range of values between minimum and typical. The minimum available C/kT due to these various uncertainties is 54.2 dB-Hz for two carrier operation. This results in an estimated 2.9 dB minimum margin over the required C/kT value* of 51.3 dB-Hz needed to realize the experimental digital radio objectives at a design point BER of 10^{-3}. For meaningful system testing under simulated realistic conditions, the extra margin can be reduced as needed by lowering the up-link output power. Reduction of up-link power is accomplished simply by reducing the power amplifier power supply voltage.

For comparative purposes, operation of FM-SCPC is also possible down to a threshold C/kT of about 46 dB-Hz, although nominal operation is typically 51 dB-Hz. The feasibility of FM operation at low C/kT values can be evaluated during the analog FM voice communication experiment, in conjunction with the testing of the digital radio system. Note that for FM systems the threshold and the end of the usable range occur at essentially the same point.

*Using BPSK with CVSD at 16 kbps, operation is possible down to the end of the usable intelligibility range at about 43.6 dB-Hz including 2.5 implementation margin.

Table 9.4-3. Digital Radio System Parameters and Link Budgets

UP-LINK FREQUENCY	1630 TO 1650 MHz
UP-LINK:	
EIRP OF EARTH TERMINAL	16.0 dBW
PATH LOSS TO SATELLITE	189.2 dB
C/kT	59.7 dB-Hz
DOWN-LINK:	
EIRP OF SATELLITE	50.1 dBW
PATH LOSS TO EARTH TERMINAL	188.7 dB
C/kT	63.0 dB-Hz (1 CARRIER)
COMBINED C/kT FOR 1 CARRIER	58.0 dB-Hz
COMBINED C/kT FOR 2 CARRIERS	56.8 dB-Hz

UNCERTAINTIES

UP-LINK:	
POWER AMPLIFIER GAIN	-0.5 dB
MISMATCH LOSS VARIATIONS	-0.9 dB
OUTPUT FILTER BANDPASS FLATNESS	-0.5 dB
ANTENNA OMNIDIRECTIVITY	-1.0 dB
UP-LINK EIRP UNCERTAINTIES	-2.9 dB
UP-LINK EIRP RANGE:	
EIRP (TYP)	16.0 dBW
EIRP (MIN)	13.1 dBW
DOWN-LINK:	
ANTENNA OMNIDIRECTIVITY	-1.0 dB
MISMATCH LOSS VARIATIONS	-0.8 dB
INPUT FILTER BANDPASS FLATNESS	-0.5 dB
DOWN-LINK G/T UNCERTAINTIES	-2.3 dB
DOWN-LINK G/T RANGE	-27 TO -29.3 dB/^{o}K
RESULTING C/kT RANGE (FOR 2 CARRIERS)	
UP-LINK	56.8 TO 59.7 dB-Hz
DOWN-LINK	57.7 TO 60.0 dB-Hz
SYSTEM	54.2 TO 56.8 dB-Hz

Table 9.4-3. Digital Radio System Parameters and Link Budgets (Continued)

LINK BUDGET:

	DIGITAL		ANALOG	
MODULATION TECHNIQUE	BPSK	QPSK	FM SCPC	
VOICE DIGITALIZATION TECHNIQUE	CVSD	CVSD	N/A	
INFORMATION DATA RATE	16	16	(18 kHz Bw)	kbps
MODULATION DESIGN E_b/N_o	6.8	6.8		dB
REQUIRED THEORETICAL C/kT	48.8	48.8		dB-Hz
IMPLEMENTATION LOSS	2.5	3.0		dB
REQUIRED PRACTICAL C/kT	51.3	51.8	51 (NOMINAL)	dB-Hz
AVAILABLE LINK C/kT (MINIMUM)	54.2	54.2	54.2	dB-Hz
DESIGN POINT LINK MARGIN	2.9	2.4	3.2	dB
ADDITIONAL AT THRESHOLD	2.5	2.5	5.0**	dB
THRESHOLD LINK MARGIN	5.4	4.9	8.2	dB
ADDITIONAL AT USABLE END POINT	5.2	5.2		dB
USABLE END POINT LINK MARGIN*	10.6	10.1	8.2	dB

*THE ACHIEVABLE USABLE END POINT MARGINS TBD DURING THE QUALITY/ INTELLIGIBILITY ASSESSMENTS

**THE INTELLIGIBILITY LIMIT (CALLED THRESHOLD HERE) OF THE FM SCPC EQUIPMENT IS 46 dB-Hz.

9.4.2 Description of Field Test Antenna. The proposed antenna for the field tests is basically an omnidirectional, bent monopole, turnstile antenna. Figure 9.4-1 illustrates the basic antenna physical configuration and Table 9.4-4 presents the antenna's electrical specifications. The manufactor of this antenna is Sensor Systems, Inc.

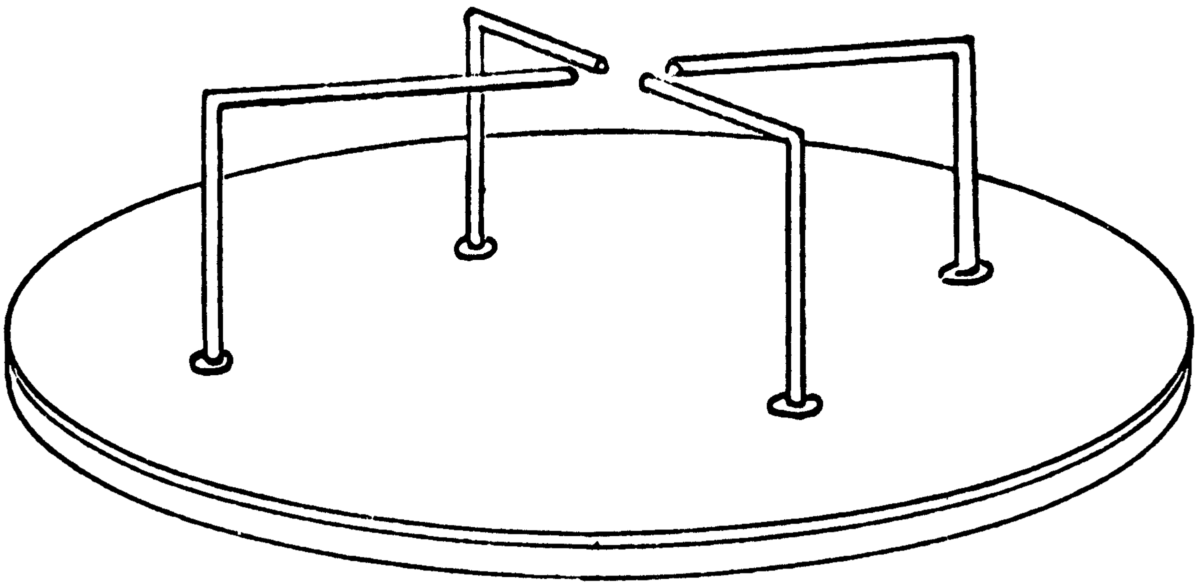


Figure 9.4-1 Bent Monopole Turnstile Antenna (Mounted on Microstrip Feed Network Without Radome Cover)

Table 9.4-4. Transmit/Receive Antenna Specifications

TYPE OF ANTENNA	BENT MONOPOLE, TURNSTILE
FREQUENCY RANGE	1535 - 1670 MHz
FIELD OF VIEW (FOV) BEAM COVERAGE	10° to 170°, HEMISPHERIC
GAIN IN FOV	1.8 dB MINIMUM
POLARIZATION	RIGHT CIRCULAR
AXIAL RATIO	1 dB, 25° to 155° 2 dB, 15° to 165°
OMNIDIRECTIVITY IN AZIMUTH	1 dB
IMPEDANCE	50 OHMS
VSWR	2:1 MAXIMUM
POWER HANDLING CAPACITY	60W MAXIMUM
MECHANICAL MOUNTING	FIRMLY MOUNTED ON TOP OF VEHICLE ABOVE RECTANGULAR GROUND PLANE WITH DIMENSIONS TBD
NOISE TEMPERATURE	250°K MAXIMUM (ASSUMED)

Desirable features of this antenna, are the almost complete hemispherical coverage (10 to 170^{o}) with uniform 2 dBI gain and a simple structural form, yielding relatively easy reproducibility and low prices in volume quantities. Although this antenna is not an off-the-shelf item and it takes NRE for development, it is a most desirable choice for the field test objectives. Antennas of this type have been built and deployed. (One model is presently being evaluated as an airborne antenna in the NAVSTAR Global positioning system.) The bent monopole, turnstile antenna will provide the desired low-angle coverage and circular polarization. In addition, the antenna may be covered simply with a hemispherical plastic radome for protection against mechanical damage or corrosion. A typical measured radiation pattern of this antenna is illustrated in Figure 9.4-2.

9.4.3 RF Unit Functional Description. A detailed functional description of the digital radio RF unit is shown in Figure 9.4-3. The transmit and receive RF chains interface with the common antenna via a broadband ferrite diplexer at L-band. The modem interface frequency is 70 MHz; this frequency was chosen because it is considered to be a standard IF for signal processing and test instrumentation.

Local oscillators are provided both in the transmit and receive chain for conversion between the 70 MHz IF and L-band. By utilizing coarse tuning (screw driver adjustment) combined with fine tuning via an AFC tuning voltage, any center frequency can be selected between 1630 - 1670 MHz for the up-link and 1535 - 1565 MHz for the down-link. Also the AFC may be utilized for freuency or phase locking the system, if needed.

Attenuators are provided for controlling the power levels at the modem interface. The attenuator in the transmit RF chain adjusts the proper level to the output power amplifier and controls the EIRP to some extent, while the attenuator at the receiver output adjusts the proper input level to the modem.

The mixer-preamplifiers are both integrated assemblies, providing good overall performance. In the down-conversion process, the receiver utilizes low-side LO injection for the mixer-preamplifier. However, in the transmit chain, high-side LO injection is used. This scheme results in a cleaner spectrum and

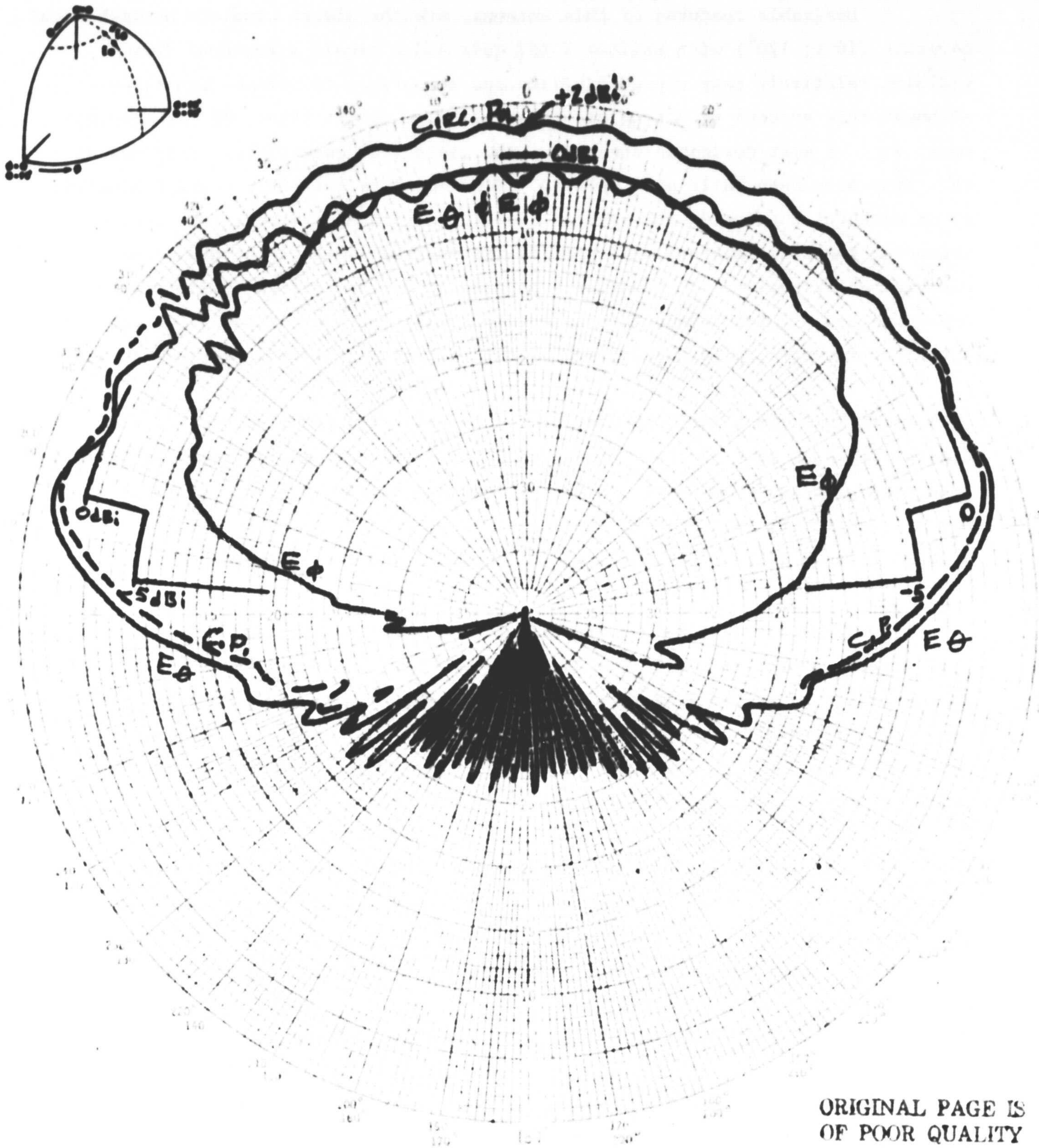


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Figure 9.4-2. Typical Polar Pattern Bent Monopole Turnstile L-Band Antenna

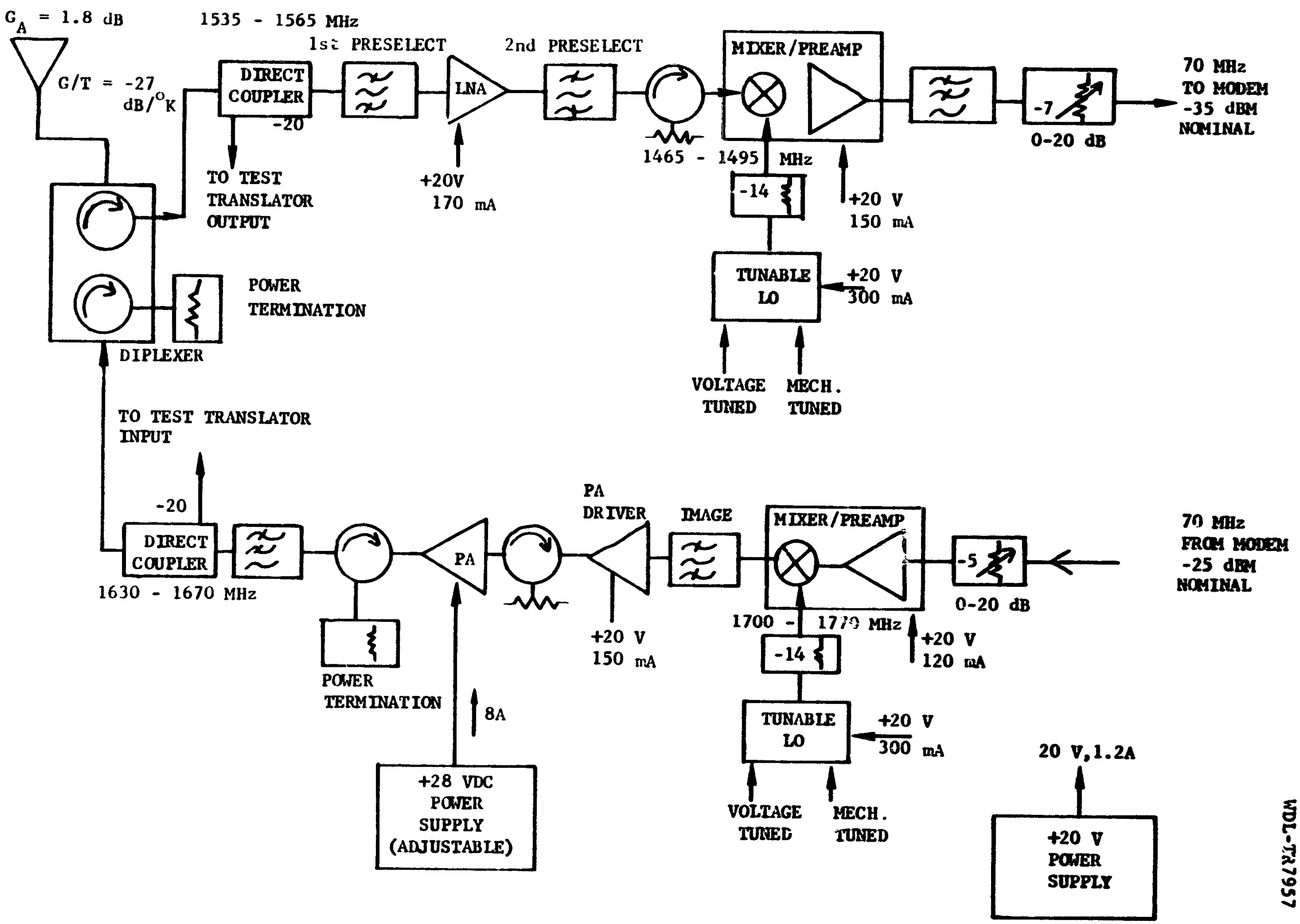

Figure 9.4-3. Digital Radio RF Unit Functional Description

filtering requirements are eased considerably, in contrast to use of low-side LO injection where the LO falls in the receive band. Thus, this design reduces the spurious levels caused by the close spacing of transmit and receive frequencies.

One bandpass filter (transmit image) and one lowpass filter (transmit output) are provided in the transmit path. Also, two bandpass filters (receive preselectors) and one lowpass filter (receive lowpass filter at 70 MHz) are provided in the receive path. The transmit image filter attenuates the image signal by 90 dB so that the leakage to the receive chain is negligible; it also attenuates the transmitted noise into the receive band that is generated due to the transmit mixer/amplifier noise.

A power amplification of 44 dB is provided utilizing a PA medium-level driver and a high power amplifier with 16 dBW output power. The power amplifier chain is all solid state and the high power amplifier operates in the saturated mode. Control of mobile terminal EIRP is accomplished conveniently by utilizing a separate, adjustable power supply for the power amplifier; viz, reduced voltage results in a reduction of EIRP. An isolator is provided at the PA input so that the PA driver sees a 50 ohm impedance instead of a poor match of the PA input. Figure 9.4-4 illustrates a commercially available high power amplifier suitable for this application.

Further attenuation of image signal and noise into the receive band takes place in the transmit output lowpass filter. This filter also attenuates the power amplifier output harmonics by 90 dB, resulting in a level of about -115 dBc maximum second harmonic level.

One purpose of the first receive preselect filter is to attenuate the transmit leakage at 1630 MHz. This signal is the strongest in the receive chain; therefore, the amplifiers were chosen with the capability to handle this high level, which is typically +3.8 dBm at the LNA output. The transmit leakage is reduced further in the second receive preselect filter, still at L-band. After mixing with the LO, and lowpass filtering the transmit leakage drops to 35 dB below the signal level. Another purpose of the receive preselector filters is to attenuate the receive image response and other spurious signals generated outside or leaking in from the transmit chain.

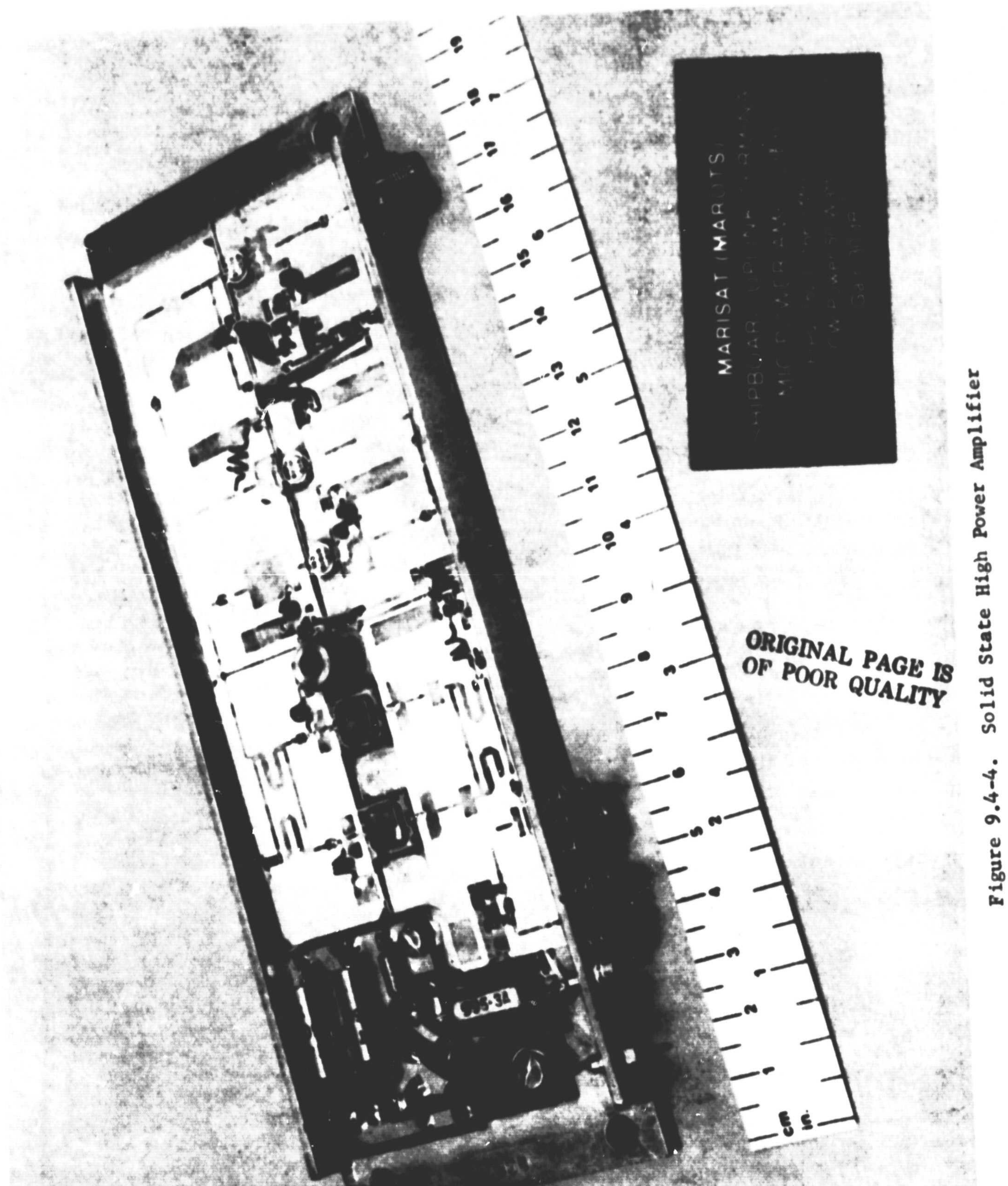

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Figure 9.4-4. Solid State High Power Amplifier

In the receive chain the low noise amplifier is a 40 dB gain amplifier having a 2.2 dB noise figure. The 1 dB compression point is +10 dBm minimum giving sufficient margin to handle the relatively high level transmit leakage signal.

The diplexer is basically a four-port wideband device, consisting of one isolator and one circulator in cascade. A 50 ohm termination on the fourth port assures good VSWR at the transmit and receive ports, regardless of mis-match at the antenna or at another port. Two 20 dB directional couplers are included at L-band on each port of the diplexer for the convenience of loop testing with a test translator.

9.4.4 Signal Level Flow and Spurious Analysis. Figure 9.4-5 and 9.4-6 are representative of the transmit and receive chains showing the results of the various signal level flow and spurious analysis necessary to ensure proper performance.

Significant factors affecting system performance were found to be system noise and spurious leakage from the transmitter into the receive path. The receiver chain operates, therefore, with several spurious carriers in the signal path. Thus, not only these signals, but also their associated intermodulation products resulting from the down-conversion process, need to be dealt with. Problems due to transmit leakage are reduced considerable by judicious choice of filtering requirements, frequency plan, and budgeting of gains and losses throughout the receiver.

From inspection of the block diagrams and signal levels, the following points are evident:

- Transmit noise leakage into the receive band is negligible producing no degradation of system C/kT due to this noise component
- Transmit leakage at the receiver input is +25 dBm, which is the highest level into the receiver, but this level is reduced to -70 dBm, or 35 dB below the wanted signal level at the 70 MHz receiver output. This requires a minimum of 116 dB isolation between transmitter and receiver output indicating the need for very effective shielding.

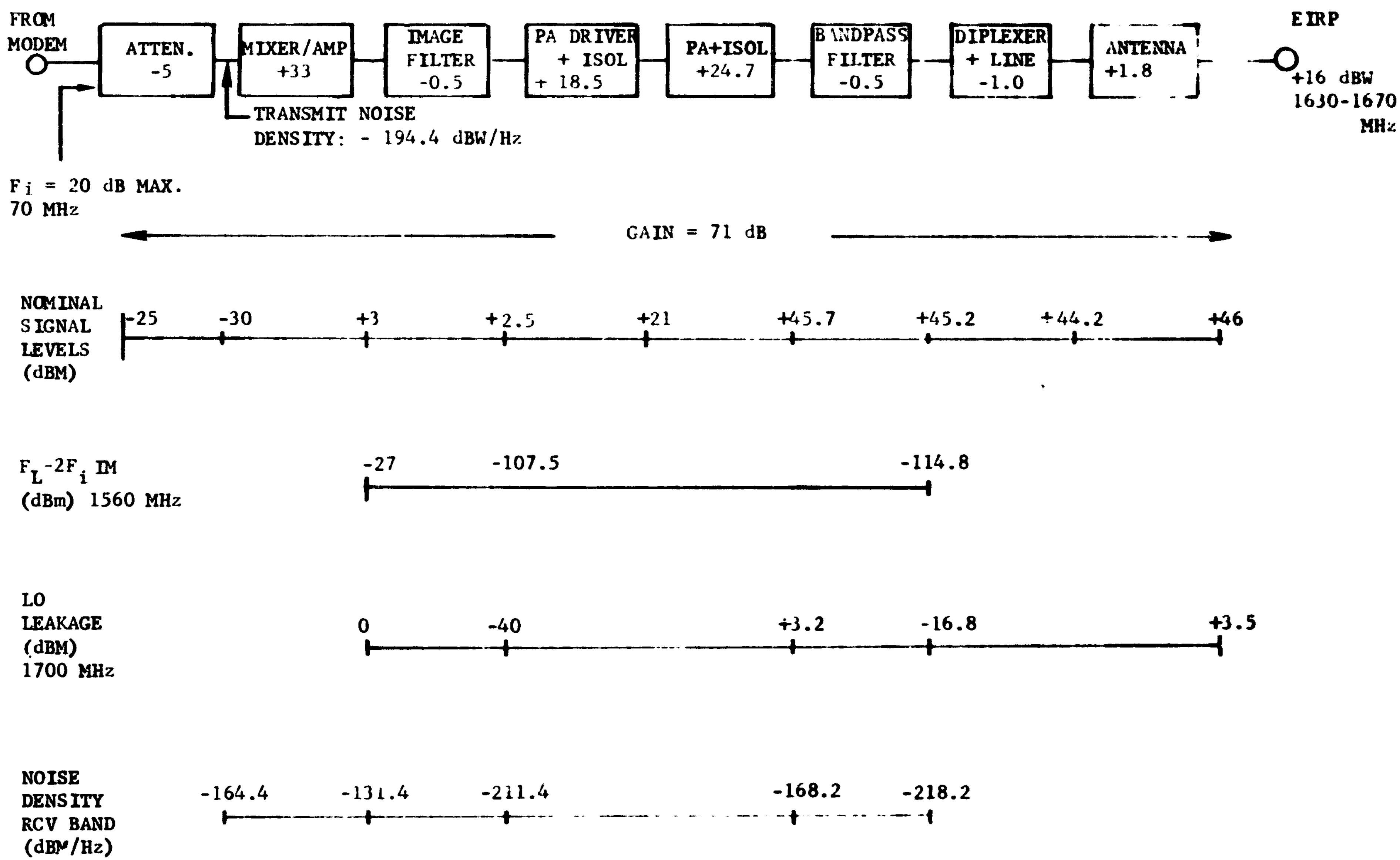

Figure 9.4-5. Transmit Chain and Level Diagram

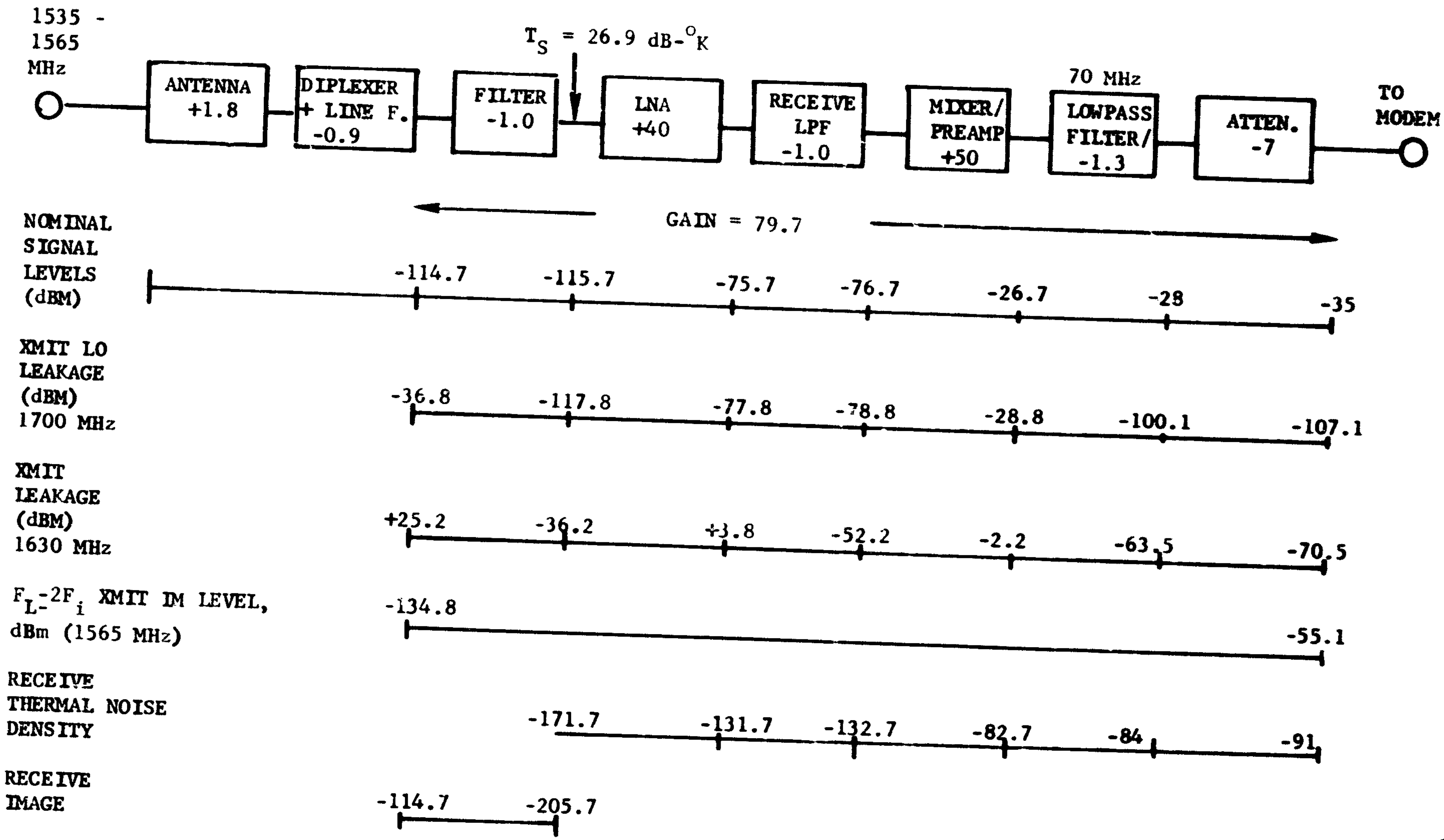

Figure 9.4-6. Receive Chain and level Diagram

- The transmit LO leakage is the next highest level signal at the receiver input, but it is easier to filter this spurious because it has wider separation from the receive carrier.
- The intermodulation product, $F_L - 2F_i$ generated during the up-conversion process falls directly into the receive band at 1560 MHz (for typical values of $F_L = 1700$ MHz and $F_i = 70$ MHz). It is important to choose the proper mixer configuration and adequate filtering in the transmit chain to suppress this product. With the present scheme, this IM product is about 20 dB below the receive signal, which is considered acceptable.

One can conclude, therefore, that the selected design scheme yields problem-free operation, provided that adequate shielding is provided between the transmitter and the receiver and that all components are selected for good performance.

9.5 Technical Description of Digital Radio Baseband Unit

Previously, the operational system description effort identified several technical parameters that are required for a digital modem intended for mobile radio applications. Table 9.5-1 highlights the key technical features and parameters identified here for the demonstration field tests.

As described previously in Section 9.2.2, a diligent canvassing of modem vendors turned up no single PSK modem possessing all of the required qualities. Since no suitable modem is available from outside sources, the only solution appears to be modification of an existing modem to achieve the required goals.*

*An applicable FM-SCPC modem was developed by FACC and utilized in the Indonesian DOMSAT program.

Table 9.5-1. Digital Baseband Unit Technical Features

ITEM	REQUIREMENTS
MODULATOR:	• BPSK operation at 16 kbps • Capable of spectral shaping to reduce sideband levels (if necessary)
DEMODULATOR:	• BPSK operation down to E_b/N_o of 3 dB or less • Less than 2.5 dB deviation from theoretical • Rapid carrier acquisition and clock recovery (3-4 msec) at the lowest E_b/N_o with frequency uncertainties as high as 3 kHz to prevent first syllable clipping • High noise immunity in the carrier detection circuits to prevent false squelching/unsquelching of the receiver at low E_b/N_o operation.
VOX OPERATION:	• Turn transmitter on when speech is present • Turn transmitter off during absence of speech, including pauses • High noise immunity on carrier activation circuits to reduce false carrier activations • No first syllable or last syllable clipping

Figure 9.5-1 details the essential functional blocks required for a BPSK mobile radio baseband subsystem. The frequency range, channel spacing, and IF chain configuration are identical to that of an available FM-SCPC radio system. Thus, the necessary modifications to the existing FM radio would require development of the following items:

a. BPSK modulator (minor)

b. Delta modulation coder/decoder (minor

c. BPSK demodulator (moderate to difficult, depending upon E_b/N_o operating point)

d. Bit synchronizer (moderate to difficult, depending upon E_b/N_o operating point)

e. Digital control unit (moderate since DAMA not to be implemented)

f. Digital shaping (minor to difficult depending upon sophistication required)

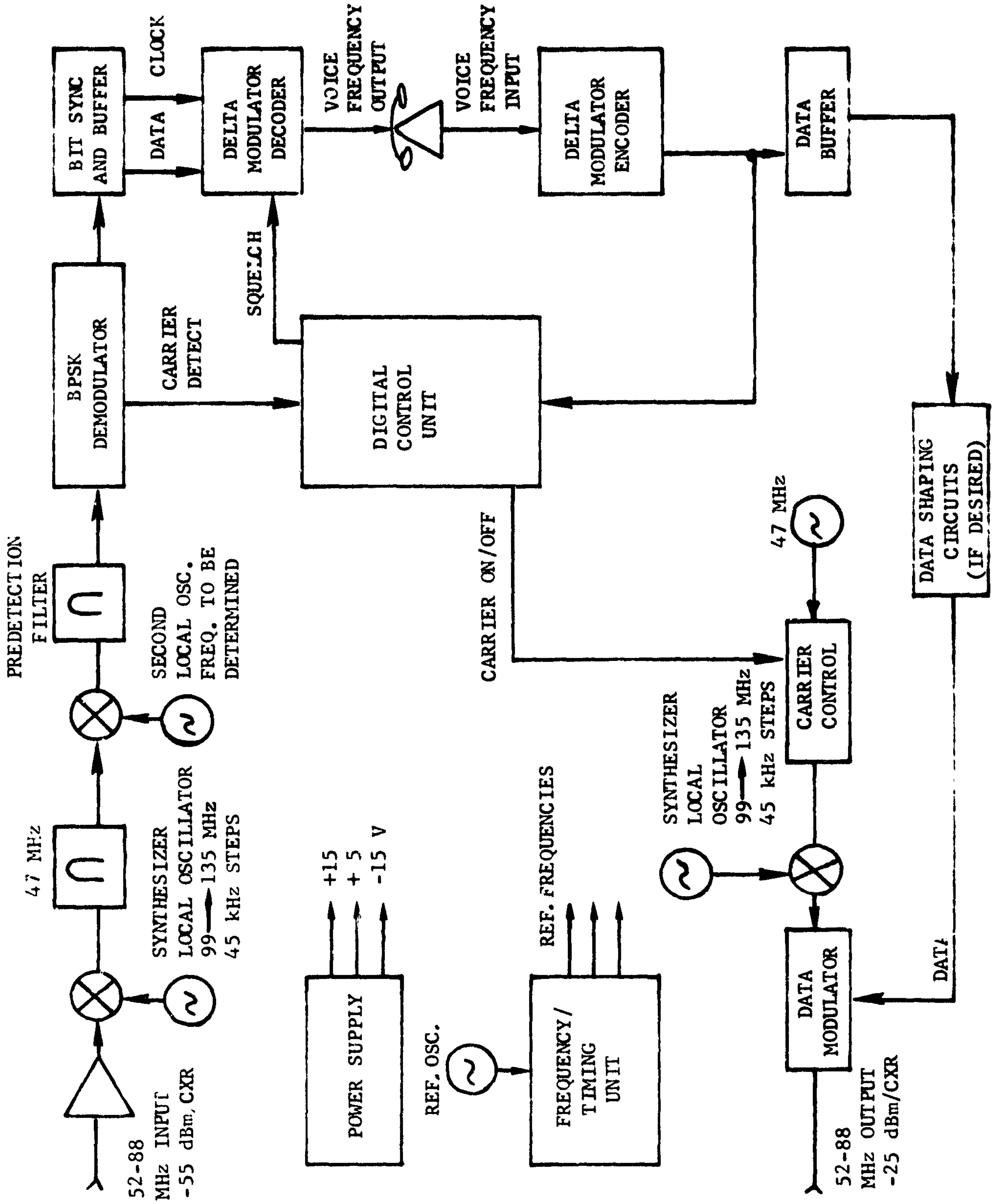

Figure 9.5-1. Digital Radio Baseband Unit

Associated with each of these modification items is a comment regarding the level of complexity attributed to achieving these revisions. Some of these modifications for digital operation already have been under development and it appears quite feasible to work in conjunction with the FM radio system. However, it should be evident that the major portion of this modem already exists in operational form and includes both the up-and the down-converter chains along with all of the frequency synthesizers, filters, mixers, and associated dircuitry.

A serendipity attributed to working with a modified FM radio system is that it would also allow comparison of performance between FM and digital modulation under various controlled link conditions. While this approach requires non-recurring costs, it does meet the objectives of the experiment plan and provides the means to prove that the technical features listed in Table 9.5-1 can be achieved. In summary, it appears that this approach provides the valuable opportunity to achieve two interesting objectives with essentially one piece of equipment.

9.6 Summary of Proposed Tests

This section highlights the types of tests and test procedures involved in the field demonstration experiment, but is not intended to be either an exhaustive or a detailed description of the proposed tests.

9.6.1 Link Margins/System Parameters. Prior to conducting any baseband testing, RF tests should be conducted to verify performance of the transmit and receive RF chains, first individually, and then in tandem in a loop test utilizing a test translator. Specifically, the following performance data need be obtained for the basic RF equipment:

a. Up-link EIRP
b. Antenna polar gain patterns in 2 planes
c. Figure of Merit, G/T and available C/kT
d. Link margins (RF portion)
e. Transmit/Receive chain signal/noise levels

The above parameters are to be measured either individually or on a loop basis, as appropriate, prior to field testing to establish reference levels.

9.6.2 Intelligibility, Quality vs E_b/N_o and BER. These parameters, relating to the baseband equipment, are tested following verification of the RF (70 MHz to 70 MHz) performance. The testing will be performed with a BPSK modem operating at 16 kbps. The following parameters need to be tested:

a. Informal assessment of intelligibility and quality at design point C/kT
b. Determination of threshold link margin by decreasing power
c. Measurement of BER versus C/kT using BPSK
d. Determination of usable end point link margin
e. Verification of modem performance (acquisition times, lock-up threshold, etc)
f. Other factors that may be found detrimental to performance

All of the above tests are performed in the laboratory prior to field tests, with conditions in the lab closely simulating conditions to be encountered in the field for realistic evaluation and to establish a reference point for the digital equipment. Measurement of C/kT will be performed by injecting noise from a suitable noise generator at the 70 MHz interface using a power combiner and divider together with a spectrum analyzer.

9.6.3 FM Analog Voice Communication Tests. To perform this experiment, an FM-SCPC modem is required. Since the FM system appears to be usable down to a low C/kT of about 46 dB-Hz, it is considered worthwhile to investigate narrow-band FM performance within this effort. The following are the parameters to be measured in the FM analog voice communication experiment.

a. C/kT versus output signal-to-noise ratio
b. Subjective quality at C/kT = 51.3 dB-Hz, which is the required value to realize the design value BER for digital operation
c. Acquisition time and lock-up threshold C/kT
d. Usable end point FM threshold

As in the case of digital operation, the parameters in b and d above are to be determined using subjective evaluation. Pretesting of above parameters is to be performed in the lab prior to testing in the field.

9.6.4 Field Evaluation and Testing. In the field testing phase the following parameters are considered for evaluation for BPSK operation.

a. Quality assessment and C/kT
b. Earth - Satellite - Earth link margins
c. Threshold margin
d. Usable end point margin
e. VOX operation

For FM analog operation, the same parameters should be measured, but utilizing test tone-to-noise ratios. In the case of FM, the degradation is rather sharp below threshold, so the usable end point margin and threshold margin may coincide.

During field tests, many different conditions are present due to terrain irregularities, local interferences, multipath losses etc. The scope of this task includes evaluation of the above parameters for one set of conditions only. These field conditions will be defined at the beginning of the testing phase.

9.7 Field Experiment Cost Estimate

The cost estimates for carrying out the objectives of the field demonstration experiment are shown in itemized form in Table 9.7-1. These costs were developed on the following baseline consisting of three phases.

PHASE 1:

All equipment are built to conformance corresponding to the cost schedule of items listed in the cost breakdown of Table 9.7-1. Two FM-SCPC baseband units are available that can be modified to meet the requirements of the field testing for the NRE shown: these units can be loaned to NASA for the RE cost figure shown for the duration of the field tests (3 to 4 months). Note that, the baseband units are to be modified to have the capability to perform not only the digital BPSK, but also the FM analog communication experiment. As an option, these baseband units can be purchased by NASA for an additional sum of $14,000.

Table 9.7-1. Field Experiment Cost Breakdown

PHASE 1	RECURRING EXPENSES	NON-RECURRING EXPENSES
ITEMS:		
1. HARDWARE COST FOR 2 RF UNITS	31,500	13,500
2. HARDWARE COST FOR 2 BASEBAND UNITS	6,000 (OPTION: 20,000)**	50,000
3. HARDWARE COST FOR 1 TEST TRANSLATOR	5,500	3,300
4. OPERATIONAL CHECKOUT, RADIO, RF	1,800	1,000
5. OPERATIONAL CHECKOUT, TEST TRANSLATOR	800	
6. SUBCONTRACTS/PROCUREMENT	3,700	
7. TRAVEL EXPENSES		1,000
SUBTOTAL:	$ 49,300	$ 68,800
TOTAL:	$118,100	
TOTAL WITH OPTION:	$132,100	

PHASE 2	EXPENSES
ITEMS:	
1. TEST EQUIPMENT COSTS	5,000
2. PLANNING/SUPERVISION	1,600
3. TEST ENGINEERING	6,000
TOTAL:	$ 12,600

*BASEBAND UNIT INCLUDES: 2 POWER SUPPLIES
2 MODEMS
2 FREQUENCY/TIME UNITS
2 OSCILLATORS

**OPTION: TWO COMPLETE BASEBAND EQUIPMENTS ARE SOLD TO NASA AT $14,000 ABOVE RECURRING COST FOR LOANED EQUIPMENT.

Table 9.7-1. Field Experiment Cost Breakdown (Continued)

PHASE 3	EXPENSES
ITEMS:	
1. PLANNING OF EXPERIMENTS, TASK DEFINITION, TEST PROCEDURES	4,700
2. TEST EQUIPMENT	5,000
3. TRAVEL	3,000
4. TEST ENGINEERING (2 MAN MONTHS)	16,600
5. SUPERVISION	5,000
6. FINAL REPORT	2,500
7. SATELLITE SCHEDULING	1,000
8. LAND MOBILE VEHICLE LEASE	1,000
TOTAL:	$ 38,800

TOTAL, PHASE 1, 2, and 3	$169,500
MISCELLANEOUS (ODC, ETC.)	$ 500
GRAND TOTAL	$170,000

PHASE 2:

All equipment to be tested under laboratory conditions to the extent necessary to verify that the equipment will operate satisfactorily in the field. In Phase 2, final adjustments and perhaps minor changes will be made, if needed to align and prepare the units for PHASE 3, where the actual field tests take place.

Two major cost items are involved; the test engineering and test equipment costs. The test engineering costs are fairly well fixed. The test equipment costs, however, are somewhat flexible. Costs shown for the test equipment is based on leasing about 80% of the equipment.

PHASE 3:

Costing for PHASE 3 is approximate because complete details are not available and the field test objectives are defined only in generalities. Thus, basically PHASE 3 is somewhat open and should be more closely defined in a detailed manner that reflects a more realistic cost estimate. However, for the sake of completeness, eight possible cost items are listed. These cost figures are based on experience gained by participation in similar hardware programs.

The baseline employed here utilizes the same in-lab testing for many of the same parameters as in PHASE 2, but under a different set of test conditions, since PHASE 3 tests are performed in the field, utilizing the satellite. Test results certainly will be influenced by this; for example, satellite position and multipath effects are just two of the many conditions directly affecting performance. The cost model for PHASE 3 is based on testing parameters under only one set of conditions. These test conditions have to be defined in detail and agreed to prior to the beginning of the task; until then the costs presented in PHASE 3 must be considered tentative.

Satellite scheduling includes only the administrative effort required by the test and supervisory personnel to secure a time slot for experimenting with the ATS-6 satellite. The costs of satellite programming, control, and usage fees are assumed to be fully covered by NASA.

The test engineering expenses include straight hours with no per diem, food, lodging, or other miscellaneous expenses. Depending on the assigned location during the field testing phase, these expenses should be added to the total costs of PHASE 3, if necessary.

As shown in the cost breakdown, using 1978 prices and wages, the grand total sum for carrying out the field demonstration experiment plan is estimated to be about $170,000. This figure does not include the costs for general administrative functions and overhead which are peculiar to the organization carrying out the experiment. If the option to purchase the FM-SCPC modems is exercised, an additional amount of $14,000 must be added for a total of $184,000.

The expected time to complete the experiment plan is estimated to be about 8 to 11 months.

9.8 Quantitative Intelligibility and Quality Testing

The plan for the demonstration experiment provides for measurements of various system and equipment parameters such as available power, acquisition times, and so on. While these measurements are very important for evaluating and assessing the system concept, and for determining if the equipment is operating as it was designed, it will not be possible to derive from them a quantitive figure of merit for voice intelligibility and quality. The only reliable, repeatable method of quantitatively comparing voice communication systems involves the use of human listeners to evaluate actual transmissions. This study has presented intelligibility and quality data for some digital voice coding methods (mainly CVSD) as a function of bit error rate. The performance was then related to available power, but under ideal conditions. In a real field system, the ideal values of BER for a given available power may not be achieved, and thus, it is worthwhile to assess voice performance under actual conditions. In addition, quantitative data has not been available on voice performance for analog FM systems. Hence, this section proposes

an extension to the current field experimental plan that includes quantitative comparative testing of voice performance for digital and analog configurations.

This approach is two-fold and is aligned with the primary proposed demonstration plan. It consists of evaluating the voice performance of the experimental configuration using standardized methods during the laboratory phase, where transmission conditions can be precisely controlled, and then verifying this performance in the field experiment phase under actual operating conditions. During the laboratory phase, naive users with no previous exposure to these communication systems will be used in order to obtain comparable results with existing studies. During the field phase, trained speech listeners will talk over the candidate voice systems in order to verify that the performance is as predicted during the laboratory phase, and that all actual operational conditions are accounted for that affect voice performance.

The planned rating tests would include assessment of intelligibility and quality, using standard prepared tapes and panels of human listeners in addition to a two-way interactive task-completion experiment where the voice transmission system under test is used for communicating instructions. In the first set of tests, tapes are run through the candidate communication system and recorded at the receiver output. These recorded tapes are then presented to a panel of human listeners for evaluation. In the second test, the systems are ranked according to total amount of information exchanged, time to complete the task, ease of communication, and so on. It should be noted that these tests are designed to measure independant items. The intelligibility and quality tests measure the types of degradation that occur, how they affect what is said, and how it sounds. The two-way tests attempt to further quantify the degradation and determine whether users compensate for the degradation by adaptation. It is well-known that this can and does occur in actual use of communication systems and explains why certain systems can be more servicable than predicted from the intelligibility scores. In view of the operational power ranges expected in the mobile terminal environment, the two-way evaluations are considered very important.

The voice coding techniques that are candidates for further testing are those that appear most promising from a cost and performance viewpoint and include CVSD, ADM (specifically the Deltamodulation Inc. version), and analog FM. The digital systems should be evaluated at data rates from 9.6 to 32 kbps, and over a range of BER (by adjusting the available power). This will translate into reducing the received SNR for the FM system.

The cost of carrying out this type of experiment has not been determined yet, but the cost factors involved in performing these tests include the following.

- Hardware modifications (to provide the variable data rate)
- Implementation and interface costs
- Additional test equipment (for voice recordings, etc.)
- Planning and supervision of the voice tests
- Test tape preparation and running through the various systems
- Listener panel evaluations
- Test subject training and experimentation
- Field testing by qualified listeners
- Evaluation and recommendations
- Documentation

The exact amount of these costs would be determined through mutual interaction and discussions concerning the degree of interest and the exact extent of the desired testing.

With limited funds there is a lower cost option that does not require and hardware modifications to provide for the variable data rates and other voice encoding methods in the existing demonstration configuration. For this limited experiment case, the digital coding evaluation will be restricted to one rate, namely 16 kbps. The comparison for other data rates would still require extrapolation using the ideal power/BER curves, but the 16 kbps data should establish the validity of that extrapolation. The available existing body of data on intelligibility and quality can then be used for more general comparisons for other different data rates and coding techniques.

As before, the laboratory configuration can be used for performance evaluation and the field configuration for verification. Whereas the hardware portion of the costs will be reduced considerable for this type of limited testing, the overall test costs will not be reduced appreciably. Most of the costs for as few experimental conditions as exist here (in either the variable rate or single rate case) are in the experiment setup, the data analysis, and in the documentation.

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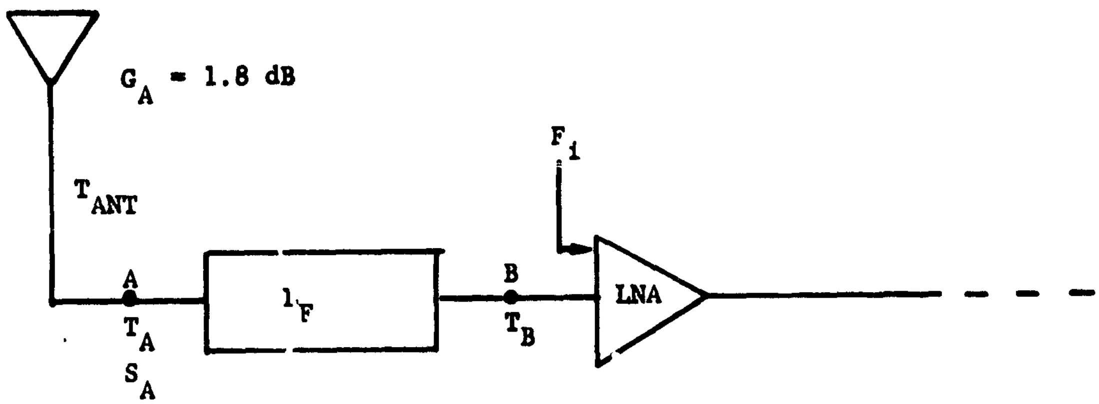
APPENDIX A

RECEIVE CHAIN FIGURE OF MERIT, G/T

Following is a representative block diagram:



System noise temp at point B:

$$T_B = \frac{T_{ANT}}{L_F} + T_o \frac{L_F - 1}{L_F} + (F_i - 1)\ T_o + \frac{T_o}{L_F^2}\left[\frac{S_A - 1}{S_A + 1}\right]^2 + T_L$$

where:

- T_{ANT} = (assume worst possible case) = 250°K
- L_F = Feed loss = combined loss of diplexer, line feed and bandpass filter = 1.9 dB = 1.55
- F_i = LNA noise figure (2.2 dB) = 1.66
- S_A = SWR of Antenna - Feed System, (computed from the RSS value of the following) = 3.08

S_1 = 1.25, $|\rho_1|$ = 0.20 at Antenna - Diplexer junction

S_2 = 1.5 max, $|\rho_2|$ = 0.33 at Diplexer - Filter junction

S_3 = 1.5 max, $|\rho_3|$ = 0.33 at Filter - LNA junction

$$\rho_A = \sqrt{|\rho_1|^2 + |\rho_2|^2 + |\rho_3|^2} = 0.51,$$

- $T_o = 290^{o}K$
- T_L = Post Amplifier loading - negligible
- $T_B = \frac{250}{1.55} + 290 \; \frac{0.55}{1.55} + (.66)290 + \frac{290}{2.4}\left[\frac{2.08}{4.08}\right]^2$

  $= 487^{o}K \quad (26.9\ dB\text{-}^{o}K)$
- $G_B = G_A - L_F$, where G_A = Antenna gain = 1. dB in FOV
- $G_B = 1.8 - 1.9 = -0.1$ dB

Therefore at point B:

$$\text{System Figure of Merit} = \frac{G_B}{T_B} = -0.1 -26.9 = -27\ dB/^{o}K$$

(computed for LNA noise figure of 2.2 dB.)

Thermal noise density at Point B:

$$\sigma_B = kT_B = -171.7\ dBm/Hz$$

Note that the transmit noise leakage into the receiver is negligible.

GLOSSARY OF TERMS AND NOTATION

A/D	Analog to Digital
ADM	Adaptive Delta Modulation
ADPCM	Adaptive Differential Pulse Code Modulation
AFC	Automatic Frequency Control
APC	Adaptive Predictive Coding
BCC	Broadcast Channel
BER	Bit Error Rate
BPSK	Biphase Phase Shift Keying
CCD	Charge-Coupled Device
CCITT	Committee Consultant International Teleponie et Telegraphie
C/kT or C/N_o	Carrier Power to Noise Power Spectral Density Ratio
Codec	Coder-Decoder
CONUS	Continental United States
CPU	Central Processing Unit
CSC	Common Signaling Channel
CTRL	Control
CVSD	Continuously Variable Slope Delta Modulation
DAMA	Demand Assignment Multiple Access
DC	Direct Current
DCDM	Digitally Controlled Delta Modulation
DCU	Digital Control Unit
DM	Delta Modulation
DMCVS	Digital Mobile Communication Via Satellite
DoD	Department of Defense
DPCM	Differential Pulse Code Modulation
DPSK	Differentially Coherent Phase Shift Keying
DRT	Diagnostic Rhyme Test
EIRP	Effective Isotropically Radiated Power
E_b/N_o	Energy per Bit to Noise Power Spectral Density Ratio
ET	Earth Terminal
FDMA	Frequency Division Multiple Access
FEC	Forward Error Correction

GLOSSARY OF TERMS AND NOTATION (Continued)

FEMES	Fire, Emergency Medical, and Emergency Services
FM	Frequency Modulation
FOV	Field of View (3 dB Beamwidth)
FSK	Frequency Shift Keying
G/T	Receiver (Antenna) Gain to System Noise Temperature Ratio
HPA	High Power Amplifier
IC	Integrated Circuit
ID	Identification
IF	Intermediate Frequency
IM	Intermodulation (Crossproduct)
I&Q	Inphase and Quadrature
LNA	Low Noise Amplifier
LO	Local Oscillator
LPC	Linear Predictive Coding
LPF	Low Pass Filter
LSI	Large Scale Integrated
MBA	Multiple Beam Antenna
MCS	Master Control Station
Modem	Modulator-Demodulator
MRT	Modified Rhyme Test
MSK	Minimum Shift Keying
NRE	Non-Recurring Expense
ODC	Other Direct Costs
OSC	Oscillator
PA	Power Amplifier
PARM	Paired-Acceptance Rating Measure
PCM	Pulse Code Modulation
PLL	Phase Lock Loop
PSK	Phase Shift Keying
QPSK	Quadriphase Phase Shift Keying

GLOSSARY OF TERMS AND NOTATION (Continued)

RCV	Receive
RCVR	Receiver
RE	Recurring Expense
RF	Radio Frequency
S/C	Spacecraft
SCPC	Single Channel per Carrier
SNR	Signal-to-Noise Power Ratio
TBD	To Be Determined
TTY	Teletype
TWTA	Travelling Wave Tube Amplifier
UART	Universal Asynchronous Receiver-Transmitter
UHF	Ultra High Frequency
VCO	Voltage Controlled Oscillator
VDT	Voice Digitization Techniques
VF	Voice Frequency
VOX	Voice Spurt Carrier Activation
VSWR	Voltage Standing Wave Ratio
WC	Working Channel
XMIT	Transmit
XMTR	Transmitter